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(adaptive and filter and hearing aid and feedback)

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 [Print Format](#)**1 Steady-state analysis of continuous adaptation systems in hearing aids***Siqueira, M.G.; Alwan, A.; Speece, R.;*

Applications of Signal Processing to Audio and Acoustics, 1997. 1997 IEEE ASSP Workshop on , 19-1997

Page(s): 4 pp.

[\[Abstract\]](#) [\[PDF Full-Text \(460 KB\)\]](#) **IEEE CNF****2 Feedback cancellation in hearing aids: results from a computer simulation***Kates, J.M.;*

Signal Processing, IEEE Transactions on [see also Acoustics, Speech, and Signal Processing, IEEE Transactions on] , Volume: 39 Issue: 3 , March 1991

Page(s): 553 -562

[\[Abstract\]](#) [\[PDF Full-Text \(864 KB\)\]](#) **IEEE JNL****3 Steady-state analysis of continuous adaptation in acoustic feedback reduction systems hearing-aids***Siqueira, M.G.; Alwan, A.;*

Speech and Audio Processing, IEEE Transactions on , Volume: 8 Issue: 4 , July 2000

Page(s): 443 -453

[\[Abstract\]](#) [\[PDF Full-Text \(344 KB\)\]](#) **IEEE JNL****4 Steady-state analysis of continuous adaptation systems for hearing aids with a delay cancellation path***Siqueira, M.G.; Alwan, A.A.;*

Signals, Systems & Computers, 1998. Conference Record of the Thirty-Second Asilomar Conference Volume: 1 , 1-4 Nov. 1998

Page(s): 518 -522 vol.1

[\[Abstract\]](#) [\[PDF Full-Text \(344 KB\)\]](#) **IEEE CNF**

5 Feedback cancellation in hearing aids using constrained adaptation*Kates, J.M.;*

Applications of Signal Processing to Audio and Acoustics, 1999 IEEE Workshop on , 17-20 Oct. 1999

Page(s): 231 -234

[\[Abstract\]](#) [\[PDF Full-Text \(236 KB\)\]](#) **IEEE CNF****6 Bias of feedback cancellation algorithms based on direct closed loop identification***Hellgren, J.; Forssell, U.;*

Acoustics, Speech, and Signal Processing, 2000. ICASSP '00. Proceedings. 2000 IEEE International Conference on , Volume: 2 , 5-9 June 2000

Page(s): II869 -II872 vol.2

[\[Abstract\]](#) [\[PDF Full-Text \(256 KB\)\]](#) **IEEE CNF****7 Bias of feedback cancellation algorithms in hearing aids based on direct closed loop identification***Hellgren, J.; Urban, F.;*

Speech and Audio Processing, IEEE Transactions on , Volume: 9 Issue: 8 , Nov. 2001

Page(s): 906 -913

[\[Abstract\]](#) [\[PDF Full-Text \(178 KB\)\]](#) **IEEE JNL****8 Feedback cancellation in hearing aids***Kates, J.M.;*

Acoustics, Speech, and Signal Processing, 1990. ICASSP-90., 1990 International Conference on , 3-1990

Page(s): 1125 -1128 vol.2

[\[Abstract\]](#) [\[PDF Full-Text \(280 KB\)\]](#) **IEEE CNF****9 Measurement and adaptive suppression of acoustic feedback in hearing aids***Bustamante, D.K.; Worrall, T.L.; Williamson, M.J.;*

Acoustics, Speech, and Signal Processing, 1989. ICASSP-89., 1989 International Conference on , 23-1989

Page(s): 2017 -2020 vol.3

[\[Abstract\]](#) [\[PDF Full-Text \(300 KB\)\]](#) **IEEE CNF****10 Asymmetric crosstalk-resistant adaptive noise canceller and its application in beamfo***Peng, W.; Kuo, S.-M.;*

Circuits and Systems, 1992. ISCAS '92. Proceedings., 1992 IEEE International Symposium on , Vol 3-6 May 1992

Page(s): 513 -516 vol.2

[\[Abstract\]](#) [\[PDF Full-Text \(308 KB\)\]](#) **IEEE CNF****11 Feedback cancellation in hearing aids: results from using frequency-domain adaptive***Estermann, P.; Kaelin, A.;*

Circuits and Systems, 1994. ISCAS '94., 1994 IEEE International Symposium on , Volume: 2 , 30 M June 1994
Page(s): 257 -260 vol.2

[Abstract] [PDF Full-Text (284 KB)] IEEE CNF

12 Subband adaptive filtering applied to acoustic feedback reduction in hearing aids

Siqueira, M.G.; Speece, R.; Petsalis, E.; Alwan, A.; Soli, S.; Gao, S.;
Signals, Systems and Computers, 1996. 1996 Conference Record of the Thirtieth Asilomar Conference
Volume: 1 , 3-6 Nov. 1996
Page(s): 788 -792 vol.1

[Abstract] [PDF Full-Text (440 KB)] IEEE CNF

13 Some notes on feedback suppression with adaptive filters in hearing aids

Knecht, W.G.;
Applications of Signal Processing to Audio and Acoustics, 1997. 1997 IEEE ASSP Workshop on , 19-1997
Page(s): 3 pp.

[Abstract] [PDF Full-Text (228 KB)] IEEE CNF

14 An efficient feedback cancellation for multiband compression hearing aids

Young-Cheol Park; Dong-Wook Kim; In-Young Kim;
Engineering in Medicine and Biology Society, 1998. Proceedings of the 20th Annual International Conference of the IEEE , Volume: 5 , 29 Oct.-1 Nov. 1998
Page(s): 2706 -2709 vol.5

[Abstract] [PDF Full-Text (392 KB)] IEEE CNF

15 Subband signal processing for hearing aids

Wyrsch, S.; Kaelin, A.;
Circuits and Systems, 1999. ISCAS '99. Proceedings of the 1999 IEEE International Symposium on 3 , 30 May-2 June 1999
Page(s): 29 -32 vol.3

[Abstract] [PDF Full-Text (352 KB)] IEEE CNF

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05407070 E.I. No: EIP99104865423

Title: Novel approach of adaptive feedback cancellation for hearing aids
Author: Chi, Hsiang-Feng; Gao, Shawn X.; Soli, Sigfrid D.
Corporate Source: Univ of California, Los Angeles, CA, USA
Conference Title: Proceedings of the 1999 IEEE International Symposium on Circuits and Systems, ISCAS '99
Conference Location: Orlando, FL, USA Conference Date: 19990530-19990602
E.I. Conference No.: 55489
Source: Proceedings - IEEE International Symposium on Circuits and Systems v 3 1999. p III-195 - III-198
Publication Year: 1999
CODEN: PICSDI ISSN: 0271-4310 ISBN: 0-7803-5471-0
Language: English

Title: Novel approach of adaptive feedback cancellation for hearing aids
Abstract: In this paper, a band - limited adaptive adaptive feedback cancellation algorithm for hearing aids is proposed. Utilizing the characteristics of the feedback oscillation, the algorithm provides better cancellation efficiency...

Descriptors: Adaptive filtering ; Adaptive algorithms; Oscillations; Hearing aids; Transfer functions; Speech intelligibility; Acoustic properties

7/3,K/2 (Item 2 from file: 8)

DIALOG(R)File 8:Ei Compendex(R)
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05222884 E.I. No: EIP99024558325

Title: Compact, ultra low power, programmable continuous-time filter banks for feedback cancellation in hearing aid
Author: Nair, Kavita; Harjani, Ramesh
Corporate Source: Univ of Minnesota, Minneapolis, MN, USA
Conference Title: Proceedings of the 1999 12th International Conference on VLSI Design
Conference Location: Goa, India Conference Date: 19990107-19990110
E.I. Conference No.: 49707
Source: Proceedings of the IEEE International Conference on VLSI Design 1999. IEEE Comp Soc, Los Alamitos, CA, USA. p 55-60
Publication Year: 1999
CODEN: PIVDEZ
Language: English

Title: Compact, ultra low power, programmable continuous-time filter banks for feedback cancellation in hearing aid
Abstract: This paper describes the design of a compact, ultra low power continuous time programmable filter . Compact programmable filters are required for a number of applications. A particular application is in feedback cancellation filters in hearing aids . Here, a feedback cancellation filter that matches the open loop transfer function is used to suppress acoustic oscillations. Because of the complexity of the transfer function the number of poles in the cancellation filter is fairly large. To realize an integrated cancellation filter an ultra-low power transconductance cell with large linear range has been designed. Both measurement...

...implementation and show that there is a significant savings in area as

June 27, 2003

well. The complete **filter** has been designed, implemented and fabricated in an 2 mu CMOS technology. (Author abstract) 17...

Descriptors: Digital **filters** ; Low pass **filters** ; Hearing aids; Feedback control; Transfer functions; Electric network synthesis; Transconductance; Capacitors; CMOS integrated circuits; Computer...

Identifiers: Programmable-time **filter** banks; Feedback cancellation **filters**

7/3,K/3 (Item 1 from file: 34)
DIALOG(R)File 34:SciSearch(R) Cited Ref Sci
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11280314 Genuine Article#: 628PN No. References: 29
Title: Band - limited feedback cancellation with a modified filtered -X LMS algorithm for hearing aids
Author(s): Chi HF (REPRINT) ; Gao SX; Soli SD; Alwan A
Corporate Source: Virata Corp,2700 San Tomas Expressway/Santa Clara//CA/95051 (REPRINT); Virata Corp,Santa Clara//CA/95051; House Ear Res Inst,Los Angeles//CA/90057; Univ Calif Los Angeles,Dept Elect Engn,Los Angeles//CA/90095
Journal: SPEECH COMMUNICATION, 2003, V39, N1-2 (JAN), P147-161
ISSN: 0167-6393 Publication date: 20030100
Publisher: ELSEVIER SCIENCE BV, PO BOX 211, 1000 AE AMSTERDAM, NETHERLANDS
Language: English Document Type: ARTICLE (ABSTRACT AVAILABLE)

Title: Band - limited feedback cancellation with a modified filtered -X LMS algorithm for hearing aids
...Abstract: not provide satisfactory performance for reducing feedback oscillation in hearing aids. In this paper, a **band - limited** adaptive feedback cancellation algorithm using normalized **filtered -X** LMS techniques is proposed that provides good cancellation efficiency, convergence behavior and better output...

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06238725 E.I. No: EIP02517277836

Title: Band - limited feedback cancellation with a modified filtered -X LMS algorithm for hearing aids
Author: Chi, Hsiang-Feng; Gao, Shawn X.; Soli, Sigfrid D.; Alwan, Abeer
Corporate Source: Virata Corporation, Santa Clara, CA 95051, United States

Source: Speech Communication v 39 n 1-2 January 2003. p 147-161

Publication Year: 2003

CODEN: SCOMDH ISSN: 0167-6393

Language: English

Title: Band - limited feedback cancellation with a modified filtered -X LMS algorithm for hearing aids
...Abstract: wideband adaptive feedback cancellation techniques do not provide satisfactory performance for reducing feedback oscillation in hearing aids. In this paper, a band - limited adaptive feedback cancellation algorithm using normalized filtered -X LMS techniques is proposed that provides good cancellation efficiency, convergence behavior and better output...

Descriptors: Hearing aids ; Feedback; Oscillations; Acoustic waves; Adaptive filtering; Algorithms; Computer simulation

Identifiers: Band - limited feedback cancellation

11/3,K/2 (Item 1 from file: 35)
DIALOG(R)File 35:Dissertation Abs Online
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01296851 ORDER NO: AAD93-19025

TWO-BAND QMF BANKS, BUTTERWORTH WAVELETS, TIME-VARYING FILTER BANKS AND ADAPTIVE ECHO CANCELLATION (QMF BANKS)

Author: TUNCER, T. ENGIN

Degree: PH.D.

Year: 1993

Corporate Source/Institution: BOSTON UNIVERSITY (0017)

Source: VOLUME 54/02-B OF DISSERTATION ABSTRACTS INTERNATIONAL.

PAGE 1023. 187 PAGES

TWO-BAND QMF BANKS, BUTTERWORTH WAVELETS, TIME-VARYING FILTER BANKS AND ADAPTIVE ECHO CANCELLATION (QMF BANKS)

...banks. It is shown that this method performs considerably better in terms of stopband attenuation, passband ripple, and reconstruction error compared to the time-domain formulation for the FIR case. Causal...

...biorthogonal compactly supported wavelets are constructed and compared with the Butterworth wavelets.

Echo disturbances in speaker /microphone systems is a major problem. The problems of subband echo cancellers are discussed and...

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DIALOG(R)File 2:INSPEC

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7549406 INSPEC Abstract Number: A2003-08-8770J-005, B2003-04-7520E-019

Title: Band - limited feedback cancellation with a modified filtered -X LMS algorithm for hearing aids

Author(s): Hsiang-Feng Chi; Gao, S.X.; Soli, S.D.; Alwan, A.

Author Affiliation: Virata Corp., Santa Clara, CA, USA

June 27, 2003

Journal: Speech Communication vol.39, no.1-2 p.147-61

Publisher: Elsevier,

Publication Date: Jan. 2003 Country of Publication: Netherlands

CODEN: SCOMDH ISSN: 0167-6393

SICI: 0167-6393(200301)39:1/2L.147:BLFC;1-X

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Language: English

Subfile: A B

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Title: Band - limited feedback cancellation with a modified filtered -X LMS algorithm for hearing aids

...Abstract: wideband adaptive feedback cancellation techniques do not provide satisfactory performance for reducing feedback oscillation in hearing aids . A band - limited adaptive feedback cancellation algorithm using normalized filtered -X LMS techniques is proposed that provides good cancellation efficiency, convergence behavior and better output...

...Descriptors: hearing aids ;

...Identifiers: hearing aids ;

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DIALOG(R)File 94:JICST-Eplus

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03840589 JICST ACCESSION NUMBER: 99A0045892 FILE SEGMENT: JICST-E

Howling Suppression for a Tele-Communication System using both a Time-Variant Notch Filter and an Acoustic Echo Canceller .

UCHIYAMA MICHIAKI (1); TOYAMA MIKIO (1); HIRAI TOORU (2)

(1) Kogakuin Univ.; (2) Yamaha Corp.

Denshi Joho Tsushin Gakkai Gijutsu Kenkyu Hokoku(IEIC Technical Report
(Institute of Electronics, Information and Communication Engineers),
1998, VOL.98,NO.277(EA98 59-64), PAGE.41-46, FIG.10, REF.2

JOURNAL NUMBER: S0532BBG

UNIVERSAL DECIMAL CLASSIFICATION: 621.391.8 621.372.54

LANGUAGE: Japanese COUNTRY OF PUBLICATION: Japan

DOCUMENT TYPE: Journal

ARTICLE TYPE: Original paper

MEDIA TYPE: Printed Publication

Howling Suppression for a Tele-Communication System using both a Time-Variant Notch Filter and an Acoustic Echo Canceller .

ABSTRACT: As acoustic collaboration technologies are being developed, speaker -phone control, such as howling control in a closed loop has been important. In this...

BROADER DESCRIPTORS: band stop filter...

11/3,K/5 (Item 2 from file: 94)

DIALOG(R)File 94:JICST-Eplus

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01786012 JICST ACCESSION NUMBER: 93A0619340 FILE SEGMENT: JICST-E

Control of Antiresonance by a Variable Damping Active Resonator.

MATSUHISA HIROSHI (1); SATO SUSUMU (1); TSUJIMOTO IKUO (2)

(1) Kyoto Univ., Faculty of Engineering; (2) Kyoto Univ., Graduate School
Nippon Kikai Gakkai Ronbunshu. C(Transactions of the Japan Society of
Mechanical Engineers. C), 1993, VOL.59,NO.562, PAGE.1824-1829, FIG.12,
REF.5

JOURNAL NUMBER: F0045BAL ISSN NO: 0387-5024

UNIVERSAL DECIMAL CLASSIFICATION: 534.2+534.8

LANGUAGE: Japanese COUNTRY OF PUBLICATION: Japan

DOCUMENT TYPE: Journal

June 27, 2003

ARTICLE TYPE: Original paper
MEDIA TYPE: Printed Publication

...ABSTRACT: controlled by feedback of sound pressure detected by a microphone in the resonator to a **speaker** on the wall of the resonator cavity. When the feedback gain is increased, the damping...

...to have small damping only in the vicinity of the antiresonance, a 1/3 octave **band pass filter** is set in the **feedback loop**. Theoretical and experimental investigation was carried out of duct noise, and it was found that..

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06238725 E.I. No: EIP02517277836

Title: Band - limited feedback cancellation with a modified filtered -X LMS algorithm for hearing aids
Author: Chi, Hsiang-Feng; Gao, Shawn X.; Soli, Sigfrid D.; Alwan, Abeer
Corporate Source: Virata Corporation, Santa Clara, CA 95051, United States

Source: Speech Communication v 39 n 1-2 January 2003. p 147-161

Publication Year: 2003

CODEN: SCOMDH ISSN: 0167-6393

Language: English

Title: Band - limited feedback cancellation with a modified filtered -X LMS algorithm for hearing aids
...Abstract: wideband adaptive feedback cancellation techniques do not provide satisfactory performance for reducing feedback oscillation in hearing aids. In this paper, a band - limited adaptive feedback cancellation algorithm using normalized filtered -X LMS techniques is proposed that provides good cancellation efficiency, convergence behavior and better output...

Descriptors: Hearing aids ; Feedback; Oscillations; Acoustic waves; Adaptive filtering; Algorithms; Computer simulation

Identifiers: Band - limited feedback cancellation

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TWO-BAND QMF BANKS, BUTTERWORTH WAVELETS, TIME-VARYING FILTER BANKS AND ADAPTIVE ECHO CANCELLATION (QMF BANKS)

Author: TUNCER, T. ENGIN

Degree: PH.D.

Year: 1993

Corporate Source/Institution: BOSTON UNIVERSITY (0017)

Source: VOLUME 54/02-B OF DISSERTATION ABSTRACTS INTERNATIONAL.

PAGE 1023. 187 PAGES

TWO-BAND QMF BANKS, BUTTERWORTH WAVELETS, TIME-VARYING FILTER BANKS AND ADAPTIVE ECHO CANCELLATION (QMF BANKS)

...banks. It is shown that this method performs considerably better in terms of stopband attenuation, passband ripple, and reconstruction error compared to the time-domain formulation for the FIR case. Causal...

...biorthogonal compactly supported wavelets are constructed and compared with the Butterworth wavelets.

Echo disturbances in speaker /microphone systems is a major problem. The problems of subband echo cancellers are discussed and...

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7549406 INSPEC Abstract Number: A2003-08-8770J-005, B2003-04-7520E-019

Title: Band - limited feedback cancellation with a modified filtered -X LMS algorithm for hearing aids

Author(s): Hsiang-Feng Chi; Gao, S.X.; Soli, S.D.; Alwan, A.

Author Affiliation: Virata Corp., Santa Clara, CA, USA

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Publisher: Elsevier,
Publication Date: Jan. 2003 Country of Publication: Netherlands
CODEN: SCOMDH ISSN: 0167-6393
SICI: 0167-6393(200301)39:1/2L.147:BLFC;1-X
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Language: English
Subfile: A B
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Title: Band - limited feedback cancellation with a modified filtered -X LMS algorithm for hearing aids

...Abstract: wideband adaptive feedback cancellation techniques do not provide satisfactory performance for reducing feedback oscillation in hearing aids . A band - limited adaptive feedback cancellation algorithm using normalized filtered -X LMS techniques is proposed that provides good cancellation efficiency, convergence behavior and better output...

...Descriptors: hearing aids ;

...Identifiers: hearing aids ;

11/3,K/4 (Item 1 from file: 94)

DIALOG(R)File 94:JICST-Eplus
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03840589 JICST ACCESSION NUMBER: 99A0045892 FILE SEGMENT: JICST-E
Howling Suppression for a Tele-Communication System using both a Time-Variant Notch Filter and an Acoustic Echo Canceller .

UCHIYAMA MICHIAKI (1); TOYAMA MIKIO (1); HIRAI TOORU (2)
(1) Kogakuin Univ.; (2) Yamaha Corp.

Denshi Joho Tsushin Gakkai Gijutsu Kenkyu Hokoku(IEIC Technical Report
(Institute of Electronics, Information and Communication Engineers),
1998, VOL.98,NO.277(EA98 59-64), PAGE.41-46, FIG.10, REF.2

JOURNAL NUMBER: S0532BBG

UNIVERSAL DECIMAL CLASSIFICATION: 621.391.8 621.372.54

LANGUAGE: Japanese COUNTRY OF PUBLICATION: Japan

DOCUMENT TYPE: Journal

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Howling Suppression for a Tele-Communication System using both a Time-Variant Notch Filter and an Acoustic Echo Canceller .

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BROADER DESCRIPTORS: band stop filter...

11/3,K/5 (Item 2 from file: 94)

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01786012 JICST ACCESSION NUMBER: 93A0619340 FILE SEGMENT: JICST-E
Control of Antiresonance by a Variable Damping Active Resonator.

MATSUHISA HIROSHI (1); SATO SUSUMU (1); TSUJIMOTO IKUO (2)
(1) Kyoto Univ., Faculty of Engineering; (2) Kyoto Univ., Graduate School
Nippon Kikai Gakkai Ronbunshu. C(Transactions of the Japan Society of
Mechanical Engineers. C), 1993, VOL.59,NO.562, PAGE.1824-1829, FIG.12,
REF.5

JOURNAL NUMBER: F0045BAL ISSN NO: 0387-5024

UNIVERSAL DECIMAL CLASSIFICATION: 534.2+534.8

LANGUAGE: Japanese COUNTRY OF PUBLICATION: Japan

DOCUMENT TYPE: Journal

ARTICLE TYPE: Original paper

June 27, 2003

MEDIA TYPE: Printed Publication

...ABSTRACT: controlled by feedback of sound pressure detected by a microphone in the resonator to a **speaker** on the wall of the resonator cavity. When the feedback gain is increased, the damping...

...to have small damping only in the vicinity of the antiresonance, a 1/3 octave **band pass filter** is set in the **feedback loop**. Theoretical and experimental investigation was carried out of duct noise, and it was found that...

June 27, 2003

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04252025 E.I. No: EIP95092854328
Title: Blind separation and blind deconvolution: an information-theoretic approach

Author: Bell, Anthony J.; Sejnowski, Terrence J.
Corporate Source: Salk Inst, La Jolla, CA, USA
Conference Title: Proceedings of the 1995 International Conference on Acoustics, Speech, and Signal Processing. Part 5 (of 5)
Conference Location: Detroit, MI, USA Conference Date: 19950509-19950512
E.I. Conference No.: 43559
Source: Special Sessions ICASSP, IEEE International Conference on Acoustics, Speech and Signal Processing - Proceedings v 5 1995. IEEE, Piscataway, NJ, USA, 95CH35732. p 3415-3418
Publication Year: 1995
CODEN: IPRODJ ISSN: 0736-7791
Language: English

...Abstract: By using a new algorithm, nearly perfect separation of up to 10 digitally mixed human **speakers** is achieved. This performance is significantly better than any previous algorithm. When used for deconvolution, the technique automatically **cancels echoes** and reverberations and reverses the effects of **low - pass filtering**. 9

Refs.

Descriptors: Learning algorithms; Digital signal processing; Matrix algebra; **Low pass filters**; Codes (symbols); Statistics; Calculations; Speech processing; Echo suppression; Reverberation

14/3,K/2 (Item 2 from file: 8)
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03848202 E.I. No: EIP94041267278
Title: Adaptive active noise control system with howling canceler (2-microphone transfer function ratio system)
Author: Suzuki, Seiichirou; Hayashi, Takuro
Source: Nippon Kikai Gakkai Ronbunshu, C Hen/Transactions of the Japan Society of Mechanical Engineers, Part C v 60 n 569 Jan 1994. p 169-174
Publication Year: 1994
CODEN: NKCHDB ISSN: 0387-5024
Language: Japanese

...Abstract: a howling canceler system for active noise control in a duct. The effect of howling **canceler** employing an **echo canceler** or two kinds of 2-microphone systems (one is a 2-Microphone delay system and ...

...this howling canceler, particularly utilizing the 2-Microphone transfer function ratio system, is examined. A **filtered-X** algorithm was used, because an acoustical transfer function from the secondary source to the...

...the discrete system in the low frequency range because of the low efficiency of the **speaker** in this range. In this case, the **high - pass filter** effect of the 2-Microphone howling canceler system was useful for achieving stable control. This...

...Identifiers: active noise control system; Howling canceler system; 2-microphone transfer function ratio system; Noise reduction; **Filtered x-algorithm**; **Speaker**; Fan noise; **High pass filter**

14/3,K/3 (Item 1 from file: 35)

June 27, 2003

DIALOG(R)File 35:Dissertation Abs Online
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908363 ORDER NO: AAD86-02910

**EVALUATION OF PHASE COMPENSATION FOR ENHANCING THE SIGNAL PROCESSING
CAPABILITIES OF HEARING AIDS IN SITU**

Author: PREVES, DAVID ALLAN

Degree: PH.D.

Year: 1985

Corporate Source/Institution: UNIVERSITY OF MINNESOTA (0130)

Source: VOLUME 46/12-B OF DISSERTATION ABSTRACTS INTERNATIONAL.

PAGE 4327. 203 PAGES

**EVALUATION OF PHASE COMPENSATION FOR ENHANCING THE SIGNAL PROCESSING
CAPABILITIES OF HEARING AIDS IN SITU**

A hearing aid worn in situ was treated as a feedback control loop and was found to meet the Nyquist criteria for stability. At high gain settings, acoustic...

...to recognize. Two behavioral investigations addressed the question of whether adding phase compensation to a hearing aid significantly changed recognition scores using the speech materials selected from the literature review. The first...

...impaired listeners. The second study tested whether high frequency hearing loss could be simulated by low pass filtering the taped playback of speech stimuli and whether adding phase compensation to a hearing aid resulted in confusion patterns more closely resembling that of normal hearing listeners than that produced...

14/3,K/4 (Item 1 from file: 2)

DIALOG(R)File 2:INSPEC

(c) 2003 Institution of Electrical Engineers. All rts. reserv.

4675181 INSPEC Abstract Number: A9413-4370-004, B9407-6130-006,
C9407-1250C-003

Title: A real-time isolated word recognizer for telephone input

Author(s): Yato, F.; Kuroiwa, S.; Takeda, K.; Yamamoto, S.; Owa, K.;
Shozakai, M.

Author Affiliation: KDD Kamifukuoka R&D Labs., Saitama, Japan

Journal: Journal of the Acoustical Society of Japan (E) vol.15, no.2
p.87-96

Publication Date: March 1994 Country of Publication: Japan

CODEN: JASED2 ISSN: 0388-2861

Language: English

Subfile: A B C

...Abstract: telephone network, we developed feature extraction and a word detection algorithm. These techniques use wide band - pass filter outputs which are generally employed to decide whether speech is voiced or unvoiced. To achieve a friendly interface, the system can accept user input at any time by using an echo canceller and the new word detection algorithm. Finally, the recogniser is evaluated using a large telephone voice database consisting of more than 500 speakers .

...Identifiers: wide band - pass filter outputs...

... echo canceller ;

14/3,K/5 (Item 1 from file: 94)

DIALOG(R)File 94:JICST-Eplus

(c)2003 Japan Science and Tech Corp(JST). All rts. reserv.

01877738 JICST ACCESSION NUMBER: 93A0740540 FILE SEGMENT: JICST-E

A Study on the Characteristics of 2-microphone Transfer Function Ratio

June 27, 2003

System.

HAYASHI TAKURO (1); SUZUKI SEIICHIRO (2); (2) Toshiba Corp.
Nippon Kikai Gakkai Kikai Rikigaku, Keisoku Seigyo Koen Ronbunshu, 1993,
VOL.1993, NO.B, PAGE.127-132, FIG.12, REF.6

JOURNAL NUMBER: L1497AAE

UNIVERSAL DECIMAL CLASSIFICATION: 534.2+534.8

LANGUAGE: Japanese COUNTRY OF PUBLICATION: Japan

DOCUMENT TYPE: Conference Proceeding

ARTICLE TYPE: Original paper

MEDIA TYPE: Printed Publication

...ABSTRACT: howling canceler system for active noise control in a duct.
The effect of the howling **canceler** employing an **echo canceler** or
two kinds of 2-microphone systems (one is 2-microphone delay system and
the...

...using this howling canceler, especially using 2-microphone transfer
function ratio system, is examined. A **Filtered -X adaptive filter**
algorithm was used because acoustical transfer function from the
secondary source to the evaluation microphone...

...the discrete system in the low frequency range because of the low
efficiency of the **speaker** in this range. In this case, the **high**
pass filter effect of the 2-microphone howling canceler system is
useful for achieving stable control. The...

...delay through the control system was analyzed. This time delay concerns
the causality of FIR **filter**. Some distance between detecting
microphone and secondary source to keep this causality was confirmed
from...

...DESCRIPTORS: FIR **filter** ;

...BROADER DESCRIPTORS: digital **filter** ; ...

... **filter** (signal...

... **filter** ;

14/3,K/6 (Item 2 from file: 94)

DIALOG(R)File 94:JICST-Eplus

(c)2003 Japan Science and Tech Corp(JST). All rts. reserv.

01732128 JICST ACCESSION NUMBER: 93A0427291 FILE SEGMENT: JICST-E
A Study on Adaptive Active Noise Control System with Howling Canceller.
SUZUKI SEIICHIRO (1); HAYASHI TAKURO (1)
(1) Toshiba Corp.

Nippon Kikai Gakkai Tsujo Sokai Koenkai Koen Ronbunshu(Proceedings of the
International Sessions JSME Spring Annual Meeting), 1993,
VOL.70th,NO.Pt 3, PAGE.146-148, FIG.6, REF.4

JOURNAL NUMBER: X0588AAU

UNIVERSAL DECIMAL CLASSIFICATION: 534.2+534.8

LANGUAGE: Japanese COUNTRY OF PUBLICATION: Japan

DOCUMENT TYPE: Conference Proceeding

ARTICLE TYPE: Short Communication

MEDIA TYPE: Printed Publication

...ABSTRACT: howling canceler system for active noise control in a duct.
The effect of the howling **canceler** employing an **echo canceler** or
2-microphone system was confirmed from the viewpoint of eliminating
sound pressure signal from...

...noise reduction effect of the adaptive control system using this howling
canceler is examined. A **Filtered -X algorithm** was used because an
acoustical transfer function from the secondary source to the...

June 27, 2003

...the discrete system in the low frequency range because of the low efficiency of the **speaker** in this range. In this case, the **high pass filter** effect of the 2-microphone howling canceler system is useful for achieving stable control. This...

June 27, 2003

24/3,K/1 (Item 1 from file: 8)

DIALOG(R)File 8:Ei Compendex(R)
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06238725 E.I. No: EIP02517277836

Title: Band-limited feedback cancellation with a modified filtered-X LMS algorithm for hearing aids

Author: Chi, Hsiang-Feng; Gao, Shawn X.; Soli, Sigfrid D.; Alwan, Abeer
Corporate Source: Virata Corporation, Santa Clara, CA 95051, United States

Source: Speech Communication v 39 n 1-2 January 2003. p 147-161

Publication Year: 2003

CODEN: SCOMDH ISSN: 0167-6393

Language: English

Title: Band-limited feedback cancellation with a modified filtered-X LMS algorithm for hearing aids

Author: Chi, Hsiang-Feng; Gao, Shawn X.; Soli, Sigfrid D.; Alwan, Abeer
...Abstract: wideband adaptive feedback cancellation techniques do not provide satisfactory performance for reducing feedback oscillation in hearing aids. In this paper, a band-limited adaptive feedback cancellation algorithm using normalized filtered-X LMS...

Descriptors: Hearing aids; Feedback; Oscillations; Acoustic waves; Adaptive filtering; Algorithms; Computer simulation

24/3,K/2 (Item 2 from file: 8)

DIALOG(R)File 8:Ei Compendex(R)
(c) 2003 Elsevier Eng. Info. Inc. All rts. reserv.

05407070 E.I. No: EIP99104865423

Title: Novel approach of adaptive feedback cancellation for hearing aids

Author: Chi, Hsiang-Feng; Gao, Shawn X.; Soli, Sigfrid D.
Corporate Source: Univ of California, Los Angeles, CA, USA
Conference Title: Proceedings of the 1999 IEEE International Symposium on Circuits and Systems, ISCAS '99

Conference Location: Orlando, FL, USA Conference Date: 19990530-19990602

E.I. Conference No.: 55489

Source: Proceedings - IEEE International Symposium on Circuits and Systems v 3 1999. p III-195 - III-198

Publication Year: 1999

CODEN: PICSDI ISSN: 0271-4310 ISBN: 0-7803-5471-0

Language: English

Title: Novel approach of adaptive feedback cancellation for hearing aids

Author: Chi, Hsiang-Feng; Gao, Shawn X.; Soli, Sigfrid D.
Abstract: In this paper, a band-limited adaptive feedback cancellation algorithm for hearing aids is proposed. Utilizing the characteristics of the feedback oscillation, the algorithm provides better cancellation efficiency...

Descriptors: Adaptive filtering; Adaptive algorithms; Oscillations; Hearing aids; Transfer functions; Speech intelligibility; Acoustic properties

24/3,K/3 (Item 3 from file: 8)

DIALOG(R)File 8:Ei Compendex(R)
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04729562 E.I. No: EIP97063700503

Title: Subband adaptive filtering applied to acoustic feedback reduction

June 27, 2003

in hearing aids

Author: Siqueira, M.G.; Speece, R.; Petsalis, E.; Alwan, A.; Soli, S. ; Gao, S.

Corporate Source: Univ of California, Los Angeles, Los Angeles, CA, USA
Conference Title: Proceedings of the 1996 30th Asilomar Conference on Signals, Systems & Computers. Part 1 (of 2)

Conference Location: Pacific Grove, CA, USA Conference Date: 19961103-19961106

E.I. Conference No.: 46513

Source: Conference Record of the Asilomar Conference on Signals, Systems & Computers v 1 1997. IEEE, Los Alamitos, CA, USA, 96CB36004. p 788-792

Publication Year: 1997

CODEN: CCSCE2 ISSN: 1058-6393

Language: English

Title: Subband adaptive filtering applied to acoustic feedback reduction in hearing aids

Author: Siqueira, M.G.; Speece, R.; Petsalis, E.; Alwan, A.; Soli, S. ; Gao, S.

Abstract: Acoustic feedback is a problem in hearing aids that contain a substantial amount of gain, hearing aids that are used in conjunction with vented or open molds, and in-the-ear hearing aids. Acoustic feedback is both annoying and reduces the maximum usable gain of hearing - aid devices. This paper models the time-varying acoustic feedback path for hearing aids before performing a systematic evaluation of acoustic feedback reduction techniques through perceptual experiments. It is...

Descriptors: Adaptive filtering; Feedback; Hearing aids ; Acoustic signal processing; Algorithms; Computational complexity

24/3,K/4 (Item 4 from file: 8)

DIALOG(R)File 8:EI Compendex(R)

(c) 2003 Elsevier Eng. Info. Inc. All rts. reserv.

04560676 E.I. No: EIP96110412592

Title: Calibration, optimization, and DSP implementation of microphone array for speech processing

Author: Wang, A.; Yao, K.; Hudson, R.E.; Korompis, D.; Lorenzelli, F.; Soli, S.D. ; Gao, S.

Corporate Source: UCLA, Los Angeles, CA, USA

Conference Title: Proceedings of the 1996 9th IEEE Workshop on VLSI Signal Processing

Conference Location: San Francisco, CA, USA Conference Date: 19961030-19961101

E.I. Conference No.: 45510

Source: IEEE Workshop on VLSI Signal Processing, Proceedings 1996. IEEE, Piscataway, NJ, USA. p 221-228

Publication Year: 1996

CODEN: 85PYA8

Language: English

Author: Wang, A.; Yao, K.; Hudson, R.E.; Korompis, D.; Lorenzelli, F.; Soli, S.D. ; Gao, S.

Abstract: For various audio , teleconference, hearing aid , and voice recognition applications, a microphone array is known to be an effective method to...

24/3,K/5 (Item 5 from file: 8)

DIALOG(R)File 8:EI Compendex(R)

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04533861 E.I. No: EIP96103369592

Title: High performance microphone array system for hearing aid applications

June 27, 2003

Author: Wang, A.; Yao, K.; Hudson, R.E.; Korompis, D.; Lorenzelli, F.;
Soli, S.F. ; Gao, S.
Corporate Source: UCLA Los Angeles, Los Angeles, CA, USA
Conference Title: Proceedings of the 1996 IEEE International Conference
on Acoustics, Speech, and Signal Processing, ICASSP. Part 6 (of 6)
Conference Location: Atlanta, GA, USA Conference Date:
19960507-19960510
E.I. Conference No.: 45447
Source: ICASSP, IEEE International Conference on Acoustics, Speech and
Signal Processing - Proceedings v 6 1996. IEEE, Piscataway, NJ,
USA, 96CB35903. p 3197-3200
Publication Year: 1996
CODEN: IPRODJ ISSN: 0736-7791
Language: English

Title: High performance microphone array system for hearing aid applications

Author: Wang, A.; Yao, K.; Hudson, R.E.; Korompis, D.; Lorenzelli, F.;
Soli, S.F. ; Gao, S.

Abstract: Microphone array technology has been proposed for various
audio, teleconference, and hearing aid applications. By forming a
focused beam toward the desired speech source, attenuating background
noises and...

...In this paper, we present a high performance prototype PC-based
microphone array system for hearing aid applications. Algorithms for
maximum energy criterion array weight design needed in the speech
processing mode...

Descriptors: Acoustic arrays; Hearing aids ; Microphones; Algorithms
; Signal interference; Interference suppression; Matrix algebra; Signal
filtering and prediction; Acoustic signal processing...

24/3,K/6 (Item 6 from file: 8)

DIALOG(R)File 8:Ei Compendex(R)
(c) 2003 Elsevier Eng. Info. Inc. All rts. reserv.

04528235 E.I. No: EIP96103361906

Title: Microphone array for hearing aid and speech enhancement
applications

Author: Wang, A.; Yao, K.; Hudson, R.E.; Korompis, D.; Lorenzelli, F.;
Soli, S. ; Gao, S.

Corporate Source: UCLA, Los Angeles, CA, USA
Conference Title: Proceedings of the 1996 International Conference on
Application-Specific Systems, Architectures and Processors

Conference Location: Chicago, IL, USA Conference Date:
19960819-19960821

E.I. Conference No.: 45391

Source: International Conference on Application-Specific Systems,
Architectures and Processors, Proceedings 1996. IEEE, Piscataway, NJ, USA.
p 231-239

Publication Year: 1996

CODEN: 002451

Language: English

Title: Microphone array for hearing aid and speech enhancement
applications

Author: Wang, A.; Yao, K.; Hudson, R.E.; Korompis, D.; Lorenzelli, F.;
Soli, S. ; Gao, S.

Abstract: Microphone array technology has been proposed for various
audio , teleconference, hearing aid , and voice recognition
applications. By forming a focused beam toward the desired speech source,
attenuating...

...reverberations and competing interferences. We present a prototype

June 27, 2003

PC-based microphone array system designed for **hearing aid** applications but also applicable to other tasks. In section 1, a multiple channel microphone array...

24/3,K/7 (Item 7 from file: 8)
DIALOG(R)File 8:Ei Compendex(R)
(c) 2003 Elsevier Eng. Info. Inc. All rts. reserv.

04507867 E.I. No: EIP96093340787
Title: Novel DSP system for microphone array applications
Author: Wang, A.; Yao, K.; Hudson, R.E.; Korompis, D.; Lorenzelli, F.;
Soli, S.D. ; Gao, S.
Corporate Source: UCLA, Los Angeles, CA, USA
Conference Title: Proceedings of the 1996 IEEE International Symposium on Circuits and Systems, ISCAS. Part 2 (of 4)
Conference Location: Atlanta, GA, USA Conference Date:
19960512-19960515
E.I. Conference No.: 45321
Source: Circuits and Systems Connecting the World Proceedings - IEEE International Symposium on Circuits and Systems v 2 1996. IEEE, Piscataway, NJ, USA, 96CB35876. p 201-204
Publication Year: 1996
CODEN: PICSDI ISSN: 0271-4310
Language: English

Author: Wang, A.; Yao, K.; Hudson, R.E.; Korompis, D.; Lorenzelli, F.;
Soli, S.D. ; Gao, S.
...Abstract: Among these, digital microphone array has been proposed for sonar, audio, teleconferencing, multi-media, and **hearing aid** applications left bracket 1, 2, 4, 3, 5 right bracket . A microphone array can enhance...

...sources in various applications. We present a prototype PC-based microphone array system designed for **hearing aid** applications but also applicable to other tasks. In section 1, a multiple channel microphone array...

24/3,K/8 (Item 8 from file: 8)
DIALOG(R)File 8:Ei Compendex(R)
(c) 2003 Elsevier Eng. Info. Inc. All rts. reserv.

04368207 E.I. No: EIP96033107316
Title: Development of a prototype portable binaural digital hearing aid
Author: **Soli, Sigfrid D.**
Corporate Source: House Ear Inst, Los Angeles, CA, USA
Conference Title: Proceedings of the 1995 3rd URSI International Symposium on Signals, Systems and Electronics, ISSSE'95
Conference Location: San Francisco, CA, USA Conference Date:
19951025-19951027
E.I. Conference No.: 44409
Source: Conference Proceedings of the International Symposium on Signals, Systems and Electronics 1995. IEEE, Piscataway, NJ, USA, 95TH8047. p 381
Publication Year: 1995
CODEN: 002316
Language: English

Title: Development of a prototype portable binaural digital hearing aid
Author: **Soli, Sigfrid D.**
Abstract: In order for a **hearing aid** user to perform binaural directional hearing in noisy environments, it is important to maintain at audible levels the binaural cues that would be present if the **hearing**

June 27, 2003

aid were not in place. In view of this, a wearable prototype digital hearing aid and personal computer (PC) based methods of digital design for evaluation of the effectiveness of...

...fitting algorithms, methods and accuracy of fittings, and the hardware and software comprising the binaural hearing aid are discussed, as well as the results from field trials with the portable processors.

Descriptors: Hearing aids ; Personal computers; Digital filters; Electric network synthesis; Program processors; Microprocessor chips; Signal filtering and prediction...

Identifiers: Prototype portable binaural digital hearing aid ; Digital filter design; Body worn prototype processor; Digital signal processing chip; Binaural fitting; Hearing aid equalization; Hearing loss compensation

24/3,K/9 (Item 1 from file: 65)

DIALOG(R)File 65:Inside Conferences
(c) 2003 BLDSC all rts. reserv. All rts. reserv.

01486120 INSIDE CONFERENCE ITEM ID: CN014751276

Digital Signal Processing for An Ear Leel Binaural Hearing Aid
Soli, S.

CONFERENCE: Signals, systems and electronics-International symposium; 3rd ISSE -INTERNATIONAL SYMPOSIUM-, 1995; 3rd P: 381-381 IEEE, 1995

ISBN: 0780325168

LANGUAGE: English DOCUMENT TYPE: Conference Ppaers

CONFERENCE SPONSOR: Union Radio-Scientifique Internationale

CONFERENCE LOCATION: San Francisco, CA

CONFERENCE DATE: Oct 1995 (19951) (19951)

Digital Signal Processing for An Ear Leel Binaural Hearing Aid
Soli, S.

24/3,K/10 (Item 1 from file: 2)

DIALOG(R)File 2:INSPEC
(c) 2003 Institution of Electrical Engineers. All rts. reserv.

7549406 INSPEC Abstract Number: A2003-08-8770J-005, B2003-04-7520E-019

Title: Band-limited feedback cancellation with a modified filtered-X LMS algorithm for hearing aids

Author(s): Hsiang-Feng Chi; Gao, S.X.; **Soli, S.D.** ; Alwan, A.

Author Affiliation: Virata Corp., Santa Clara, CA, USA

Journal: Speech Communication vol.39, no.1-2 p.147-61

Publisher: Elsevier,

Publication Date: Jan. 2003 Country of Publication: Netherlands

CODEN: SCOMDH ISSN: 0167-6393

SICI: 0167-6393(200301)39:1/2L.147:BLFC;1-X

Material Identity Number: C760-2002-007

U.S. Copyright Clearance Center Code: 0167-6393/03/\$30.00

Language: English

Subfile: A B

Copyright 2003, IEE

Title: Band-limited feedback cancellation with a modified filtered-X LMS algorithm for hearing aids

Author(s): Hsiang-Feng Chi; Gao, S.X.; **Soli, S.D.** ; Alwan, A.

...Abstract: wideband adaptive feedback cancellation techniques do not provide satisfactory performance for reducing feedback oscillation in hearing aids . A band-limited adaptive feedback cancellation algorithm using normalized filtered-X LMS techniques is proposed...

...Descriptors: hearing aids ;

...Identifiers: hearing aids ;

24/3,K/11 (Item 2 from file: 2)

DIALOG(R)File 2:INSPEC

(c) 2003 Institution of Electrical Engineers. All rts. reserv.

7180527 INSPEC Abstract Number: A2002-06-8736-002, B2002-03-7520E-020

Title: On application of adaptive decorrelation filtering to assistive listening

Author(s): Yunxin Zhao; Kuan-Chieh Yen; Soli, S. ; Shawn Gao; Vermiglio, A.

Author Affiliation: Dept. of Comput. Eng. & Comput. Sci., Missouri Univ., Columbia, MO, USA

Journal: Journal of the Acoustical Society of America vol.111, no.2 p.1077-85

Publisher: Acoust. Soc. America through AIP,

Publication Date: Feb. 2002 Country of Publication: USA

CODEN: JASMAN ISSN: 0001-4966

SICI: 0001-4966(200202)111:2L.1077:AADE;1-I

Material Identity Number: J001-2002-002

U.S. Copyright Clearance Center Code: 0001-4966/2002/111(2)/1077/9/\$18.00

Language: English

Subfile: A B

Copyright 2002, IEE

Author(s): Yunxin Zhao; Kuan-Chieh Yen; Soli, S. ; Shawn Gao; Vermiglio, A.

...Descriptors: hearing aids ;

24/3,K/12 (Item 3 from file: 2)

DIALOG(R)File 2:INSPEC

(c) 2003 Institution of Electrical Engineers. All rts. reserv.

6430420 INSPEC Abstract Number: A2000-02-8770J-015, B2000-01-7520E-032

Title: A novel approach of adaptive feedback cancellation for hearing aids

Author(s): Hsiang-Feng Chi; Gao, S.X.; Soli, S.D.

Author Affiliation: Dept. of Electr. Eng., California Univ., Los Angeles, CA, USA

Conference Title: ISCAS'99. Proceedings of the 1999 IEEE International Symposium on Circuits and Systems VLSI (Cat. No.99CH36349) Part vol.3 p.195-8 vol.3

Publisher: IEEE, Piscataway, NJ, USA

Publication Date: 1999 Country of Publication: USA 6 vol. (liv+565+717+568+604+647+527) pp.

ISBN: 0 7803 5471 0 Material Identity Number: XX-1999-01882

U.S. Copyright Clearance Center Code: 0 7803 5471 0/99/\$10.00

Conference Title: ISCAS'99. Proceedings of the 1999 IEEE International Symposium on Circuits and Systems. VLSI

Conference Date: 30 May-2 June 1999 Conference Location: Orlando, FL, USA

Language: English

Subfile: A B

Copyright 1999, IEE

Title: A novel approach of adaptive feedback cancellation for hearing aids

Author(s): Hsiang-Feng Chi; Gao, S.X.; Soli, S.D.

Abstract: In this paper, a band-limited adaptive feedback cancellation algorithm for hearing aids is proposed. Utilizing the characteristics of the feedback oscillation, the algorithm provides better cancellation efficiency...

...Descriptors: hearing aids

...Identifiers: hearing aids ;

June 27, 2003

24/3,K/13 (Item 4 from file: 2)

DIALOG(R)File 2:INSPEC

(c) 2003 Institution of Electrical Engineers. All rts. reserv.

5480227 INSPEC Abstract Number: A9705-4360-001, B9703-7810C-003,
C9703-5585-001

Title: Modern microphone array for hearing aid and speech processing

Author(s): Wang, A.; Yao, K.; Hudson, R.E.; Korompis, D.; Lorenzelli, F.;

Soli, S.D. ; Gao, S.

Author Affiliation: Dept. of Electr. Eng., California Univ., Los Angeles,
CA, USA

Journal: Proceedings of the SPIE - The International Society for Optical
Engineering Conference Title: Proc. SPIE - Int. Soc. Opt. Eng. (USA)
vol.2846 p.112-21

Publisher: SPIE-Int. Soc. Opt. Eng,

Publication Date: 1996 Country of Publication: USA

CODEN: PSISDG ISSN: 0277-786X

SICI: 0277-786X(1996)2846L.112:MMAH;1-A

Material Identity Number: C574-96279

U.S. Copyright Clearance Center Code: 0 8194 2234 7/96/\$6.00

Conference Title: Advanced Signal Processing Algorithms, Architectures,
and Implementations VI

Conference Sponsor: SPIE

Conference Date: 6-8 Aug. 1996 Conference Location: Denver, CO, USA

Language: English

Subfile: A B C

Copyright 1997, IEE

Title: Modern microphone array for hearing aid and speech processing

Author(s): Wang, A.; Yao, K.; Hudson, R.E.; Korompis, D.; Lorenzelli, F.;

Soli, S.D. ; Gao, S.

Abstract: For various audio , teleconference, hearing aid , and voice
recognition applications, a microphone array is known to be an effective
method to...

...Descriptors: hearing aids ;

...Identifiers: hearing aid applications

24/3,K/14 (Item 5 from file: 2)

DIALOG(R)File 2:INSPEC

(c) 2003 Institution of Electrical Engineers. All rts. reserv.

5103032 INSPEC Abstract Number: A9524-8734-012

Title: Electrode ranking of "place pitch" and speech recognition in
electrical hearing

Author(s): Nelson, D.A.; Van Tasell, D.J.; Schroder, A.C.; Soli, S. ;
Levine, S.

Author Affiliation: Clinical Psychoacoust. Lab., Minnesota Univ.,
Minneapolis, MN, USA

Journal: Journal of the Acoustical Society of America vol.98, no.4
p.1987-99

Publication Date: Oct. 1995 Country of Publication: USA

CODEN: JASMAN ISSN: 0001-4966

U.S. Copyright Clearance Center Code: 0001-4966/95/98(4)/1987/13/\$6.00

Language: English

Subfile: A

Copyright 1995, IEE

Author(s): Nelson, D.A.; Van Tasell, D.J.; Schroder, A.C.; Soli, S. ;
Levine, S.

Descriptors: hearing aids ;

24/3,K/15 (Item 6 from file: 2)

June 27, 2003

DIALOG(R)File 2:INSPEC
(c) 2003 Institution of Electrical Engineers. All rts. reserv.

03804369 INSPEC Abstract Number: A91025100

Title: **Acoustic cues for consonant identification by patients who use the Ineraid cochlear implant**

Author(s): Dorman, M.F.; Soli, S. ; Dankowski, K.; Smith, L.M.; McCandless, G.; Parkin, J.

Author Affiliation: Arizona State Univ., Tempe, AZ, USA

Journal: Journal of the Acoustical Society of America vol.88, no.5 p.2074-9

Publication Date: Nov. 1990 Country of Publication: USA

CODEN: JASMAN ISSN: 0001-4966

U.S. Copyright Clearance Center Code: 0001-4966/90/112074-06\$00.80

Language: English

Subfile: A

Author(s): Dorman, M.F.; Soli, S. ; Dankowski, K.; Smith, L.M.; McCandless, G.; Parkin, J.

Descriptors: **hearing aids** ;

24/3,K/16 (Item 1 from file: 144)

DIALOG(R)File 144:Pascal

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15430554 PASCAL No.: 02-0122403

On application of adaptive decorrelation filtering to assistive listening
ZHAO Yunxin; YEN Kuan-Chieh; SOLI Sig ; GAO Shawn; VERMIGLIO Andy
Department of Computer Engineering and Computer Science, University of
Missouri-Columbia, Columbia, Missouri 65211; Beckman Institute and
Department of ECE, University of Illinois at Urbana-Champaign, Urbana,
Illinois 61801; Human Communication Sciences and Devices Department, House
Ear Institute, Los Angeles, California 90057

Journal: The Journal of the Acoustical Society of America, 2002-02, 111
(2) 1077-1085

Language: English

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ZHAO Yunxin; YEN Kuan-Chieh; SOLI Sig ; GAO Shawn; VERMIGLIO Andy

English Descriptors: Experimental study; Speech intelligibility; **Hearing aids** ; Decorrelation; **Acoustic filters**; Acoustic signal processing;
Adaptive filters

24/3,K/17 (Item 2 from file: 144)

DIALOG(R)File 144:Pascal

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15000067 PASCAL No.: 01-0155506

Method of measuring and preventing unstable feedback in hearing aids
GAO Shawn X; SOLI Sigfrid D

Journal: The Journal of the Acoustical Society of America, 2001-04, 109
(4) p. 1283

Language: English

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Method of measuring and preventing unstable feedback in hearing aids
GAO Shawn X; SOLI Sigfrid D

English Descriptors: Instrumentation; Measuring methods; **Hearing aids** ;
Microphones; **Acoustic variables measurement**

24/3,K/18 (Item 3 from file: 144)
DIALOG(R) File 144:Pascal
(c) 2003 INIST/CNRS. All rts. reserv.

12041469 PASCAL No.: 95-0237256
Effects of hearing aids on binaural directional hearing in hearing-impaired individuals
GELNETT Donna J; NILSSON Michael J; SOLI Sigfrid D
House Ear Inst., 2100 W. 3rd St., Los Angeles, CA 90057
The 129th Meeting of the Acoustical Society of America (Washington, DC
(USA)) 1995-05-30/1995-06-03
Journal: Journal of the Acoustical Society of America, 1995-05, 97 (5)
3346-3346
Language: English

Copyright (c) 1995 American Institute of Physics

Effects of hearing aids on binaural directional hearing in hearing-impaired individuals
GELNETT Donna J; NILSSON Michael J; SOLI Sigfrid D
...spatial separation of the speech and a spectrally matched noise for 25 hearing-impaired binaural **hearing aid** users. Directional **hearing** capacity for these individuals often fell within the normal range. Unaided RTSs were elevated 3...
... that the interaural cues for binaural directional hearing are either inaudible or absent from the **hearing aid** output. Detailed analyses will be reported with respect to the type of **hearing aid**, **hearing aid** transfer function, and degree of hearing loss.

English Descriptors: Experimental study; **Hearing** ; **Hearing aids** ; **Hearing impairment**; Directivity; Speech recognition; Noise; Transfer functions; Comparative evaluations

24/3,K/19 (Item 4 from file: 144)
DIALOG(R) File 144:Pascal
(c) 2003 INIST/CNRS. All rts. reserv.

12041467 PASCAL No.: 95-0237254
Field trials of a portable prototype digital hearing aid
GELNETT Donna J; SULLIVAN Jean A; NILSSON Michael J; SOLI Sigfrid D
House Ear Inst., 2100 W. 3rd St., Los Angeles, CA 90057
The 129th Meeting of the Acoustical Society of America (Washington, DC
(USA)) 1995-05-30/1995-06-03
Journal: Journal of the Acoustical Society of America, 1995-05, 97 (5)
3346-3346
Language: English

Copyright (c) 1995 American Institute of Physics

Field trials of a portable prototype digital hearing aid
GELNETT Donna J; SULLIVAN Jean A; NILSSON Michael J; SOLI Sigfrid D
...and receivers located in left and right ear modules was built and used in a **hearing aid** field trial. Eight hearing impaired individuals with moderate to moderately severe hearing losses served as subjects. All subjects had symmetric hearing losses and were experienced binaural **hearing aid** users. Four binaural **hearing aid** algorithms were programmed into the processor for evaluation in the field trial. The algorithms all...

English Descriptors: Experimental study; **Hearing aids** ; Digital systems; Microprocessors; Algorithms; Speech recognition; Noise; Testing

June 27, 2003

24/3,K/20 (Item 5 from file: 144)

DIALOG(R)File 144:Pascal

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11959943 PASCAL No.: 95-0140407

Method of signal processing for maintaining directional hearing with
hearing aids

SOLI Sigfrid D

Journal: Journal of the Acoustical Society of America, 1995-01, 97 (1)
733-733

Language: English

Copyright (c) 1995 American Institute of Physics

Method of signal processing for maintaining directional hearing with
hearing aids

SOLI Sigfrid D

24/3,K/21 (Item 6 from file: 144)

DIALOG(R)File 144:Pascal

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11756536 PASCAL No.: 94-0627513

Wideband microphone array for hearing aid preprocessing

YAO Kung; SOLI Sigfrid D ; KOROMPIS Dan

Elect. Eng. Dept., Eng. IV, 68-113, UCLA, Los Angeles, CA 91403-1594;
House Ear Inst., Los Angeles, CA 90057; UCLA, Los Angeles, CA 91403-1594
The 128th Meeting of the Acoustical Society of America (Austin, Texas
(USA)) 1994-11-28/1994-12-02

Journal: Journal of the Acoustical Society of America, 1994-11, 96 (5)
3244-3245

Language: English

Copyright (c) 1994 American Institute of Physics

Wideband microphone array for hearing aid preprocessing

YAO Kung; SOLI Sigfrid D ; KOROMPIS Dan

...hearing and hearing impaired individuals. The feasibility of real-time
acoustic beamformers with arrays for hearing aids , and the advantages
of this scheme over conventional adaptive schemes will also be discussed.

...English Descriptors: Experimental study; Measuring methods; HEARING
IMPAIRMENT; Microphones; Signal-to-noise ratio; Constraints; Eigenvalues;
Numerical solution; HEARING AIDS ; Speech recognition

24/3,K/22 (Item 7 from file: 144)

DIALOG(R)File 144:Pascal

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11484530 PASCAL No.: 94-0322525

Norms for a headphone simulation of the Hearing in Noise Test: Comparison
of physical and simulated spatial separation of sound sources

NILSSON Michael J; SOLI Sigfrid D

House Ear Inst., 2100 West Third St., Los Angeles, CA 90057

The 127th Meeting of the Acoustical Society of America (Cambridge,
Massachusetts (USA)) 1994-06-06/1994-06-10

Journal: Journal of the Acoustical Society of America, 1994-05, 95 (5)
2994-2994

Language: English

Copyright (c) 1994 American Institute of Physics

June 27, 2003

NILSSON Michael J; **SOLI Sigfrid D**

... noise are all lower in the headphone system, attributable to the elimination of room and **speaker** effects. Improvements in SSRTs with spatial separation of the signal and masker were 6.38...

24/3,K/23 (Item 8 from file: 144)

DIALOG(R)File 144:Pascal

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11484518 PASCAL No.: 94-0322513

Method for fitting binaural hearing aids

GAO Shawn; SULLIVAN Jean; JAYARAMAN Sriram; **SOLI Sigfrid D**
House Ear Inst., 2100 W. 3rd St., Los Angeles, CA 90057
The 127th Meeting of the Acoustical Society of America (Cambridge,
Massachusetts (USA)) 1994-06-06/1994-06-10
Journal: Journal of the Acoustical Society of America, 1994-05, 95 (5)
2991-2991

Language: English

Copyright (c) 1994 American Institute of Physics

Method for fitting binaural hearing aids

GAO Shawn; SULLIVAN Jean; JAYARAMAN Sriram; **SOLI Sigfrid D**
For a **hearing aid** wearer to perform binaural sound localization and to utilize directional hearing in noisy environments, it...

... audible levels the binaural cues (i.e., interaural time and level differences) present without the **hearing aid** (s) in place. A method of achieving this **hearing aid** fitting goal for use with a prototype digital signal processing **hearing aid** has been developed. The method includes two major steps: **hearing aid** equalization (HAE) and hearing loss compensation (HLC). HAE is achieved with an FIR filter, which equalizes the amplitude and phase insertion effects of the **hearing aids** and maintains the binaural cues with the **hearing aid** (s) in place. The HAE filter coefficients are obtained from in situ probe tube measures...

... response for the HLC filter is determined from measures of electrical signal levels in the **hearing aid** circuit during threshold tests and during reference signal presentations in the sound field. The HAE...

English Descriptors: Experimental study; Measuring methods; **HEARING AIDS**; Sound sources; Noise; Signal processing; Filters; Loudness

June 27, 2003

File 16:Gale Group PROMT(R) 1990-2003/Jun 26
(c) 2003 The Gale Group
File 160:Gale Group PROMT(R) 1972-1989
(c) 1999 The Gale Group
File 148:Gale Group Trade & Industry DB 1976-2003/Jun 25
(c) 2003 The Gale Group
File 621:Gale Group New Prod.Annou.(R) 1985-2003/Jun 25
(c) 2003 The Gale Group
File 636:Gale Group Newsletter DB(TM) 1987-2003/Jun 24
(c) 2003 The Gale Group
File 88:Gale Group Business A.R.T.S. 1976-2003/Jun 24
(c) 2003 The Gale Group
File 47:Gale Group Magazine DB(TM) 1959-2003/Jun 23
(c) 2003 The Gale group
File 275:Gale Group Computer DB(TM) 1983-2003/Jun 26
(c) 2003 The Gale Group
File 570:Gale Group MARS(R) 1984-2003/Jun 26
(c) 2003 The Gale Group
File 15:ABI/Inform(R) 1971-2003/Jun 27
(c) 2003 ProQuest Info&Learning
File 98:General Sci Abs/Full-Text 1984-2003/May
(c) 2003 The HW Wilson Co.
File 674:Computer News Fulltext 1989-2003/Jun W4
(c) 2003 IDG Communications
File 9:Business & Industry(R) Jul/1994-2003/Jun 26
(c) 2003 Resp. DB Svcs.
File 370:Science 1996-1999/Jul W3
(c) 1999 AAAS
File 369:New Scientist 1994-2003/Jun W4
(c) 2003 Reed Business Information Ltd.
File 810:Business Wire 1986-1999/Feb 28
(c) 1999 Business Wire
File 484:Periodical Abs Plustext 1986-2003/Jun W4
(c) 2003 ProQuest
File 647:cmp Computer Fulltext 1988-2003/Jun W1
(c) 2003 CMP Media, LLC
File 20:Dialog Global Reporter 1997-2003/Jun 27
(c) 2003 The Dialog Corp.
File 696:DIALOG Telecom. Newsletters 1995-2003/Jun 26
(c) 2003 The Dialog Corp.
File 634:San Jose Mercury Jun 1985-2003/Jun 26
(c) 2003 San Jose Mercury News
File 553:Wilson Bus. Abs. FullText 1982-2003/May
(c) 2003 The HW Wilson Co.
File 635:Business Dateline(R) 1985-2003/Jun 27
(c) 2003 ProQuest Info&Learning

Set	Items	Description
S1	977520	(AUDIO? OR AMPLIFICATION? OR ACOUSTIC? OR HEARING) (2N) (AID? OR DEVICE? OR APPARATUS? OR APPTS) OR SPEAKER?
S2	606149	FILTER?
S3	46195	(BAND? OR HIGH? OR LOW?) () (PASS OR STOP? OR LIMIT?) OR PAS- SBAND?
S4	29710	(FEEDBACK? OR ECHO?) (3N) (LOOP? OR CANCEL? OR PREVENT?)
S5	585	S1(3N)S4
S6	0	S5(S)S2(S)S3
S7	0	RD (unique items)
S8	585	S2(5N)S4
S9	0	S8(S)S1(S)S3
S10	0	S9 NOT S7
S11	0	RD (unique items)
S12	3	S1(S)S2(S)S3(S)S4
S13	1	RD (unique items)
S14	1	S13 NOT (S10 OR S7)
S15	4481	S2(3N)ADAPT?

June 27, 2003

S16 7 S15(S)S3(S)S4
S17 0 S16(S)S1
S18 0 S17 NOT (S14 OR S10 OR S7)
S19 123 AU=(GAO, S? OR GAO S?)
S20 3 AU=(SOLI, S? OR SOLI S?)
S21 0 AU=(GHI, H? OR GHI H?)
S22 0 S19 AND S20
S23 0 S1(S) (S19 OR S20)

June 27, 2003

14/3,K/1 (Item 1 from file: 88)
DIALOG(R) File 88:Gale Group Business A.R.T.S.
(c) 2003 The Gale Group. All rts. reserv.

06352242 SUPPLIER NUMBER: 92939068

Build the frisker: Sniff out metallic contraband with this hand-held device.

Sheets, William; Graf, Rudolf F.

Poptronics, 3, 11, 21(1)

Nov, 2002

ISSN: 1526-3681 LANGUAGE: English RECORD TYPE: Fulltext

WORD COUNT: 2820 LINE COUNT: 00206

... at the collector. Components R12, C11, and C12 form a DC-blocking and low-pass **filter** network and suppress the higher frequency components. All we want is the frequency difference product...

...amplifier stage that delivers up to a few hundred milliwatts of audio to a small **speaker** mounted off the PC board.

The Frisker is designed to only sense objects within an...

June 27, 2003

16/3,K/1 (Item 1 from file: 16)
DIALOG(R)File 16:Gale Group PROMT(R)
(c) 2003 The Gale Group. All rts. reserv.

02541147 Supplier Number: 43366941 (USE FORMAT 7 FOR FULLTEXT)

FOUR-CHIP SET COMBINES FAX, DATA, VOICE: Sierra modem chip set

Electronic Engineering Times, p21

Oct 12, 1992

Language: English Record Type: Fulltext

Document Type: Magazine/Journal; Trade

Word Count: 567

... analog front end includes 12-bit A/D and D/A converters, a seventh-order low - pass receive filter , and adaptive features for near-end echo cancellation .

The SC11083 interface IC sweeps up glue logic for the AT bus, parallel to serial...

16/3,K/2 (Item 1 from file: 148)
DIALOG(R)File 148:Gale Group Trade & Industry DB
(c)2003 The Gale Group. All rts. reserv.

03526393 SUPPLIER NUMBER: 06293006 (USE FORMAT 7 OR 9 FOR FULL TEXT)
Echo cancellation for high-speed dial-up applications. (part 1 of multipart series) (includes related article on using a V.32 echo cancelling modem) (technical)

Turner, Steven E.

Telecommunications, v22, n1, p80(5)

Jan, 1988

DOCUMENT TYPE: technical ISSN: 0278-4831

LANGUAGE: ENGLISH

RECORD TYPE: FULLTEXT; ABSTRACT

WORD COUNT: 2381 LINE COUNT: 00182

... the dial-up telephone network.

* TYPES OF ECHO

CANCELERS AND

HOW THEY WORK

Data-domain echo cancelers can be either baseband or passband cancelers or a combination of both. There are two essential differences between baseband and passband cancelers. The first is the type of signal (complex and baseband or real and passband) contained in the adaptive FIR canceler filter . The second is the location and technique by which the actual cancellation (subtraction) takes place. A block diagram of a baseband echo canceler , as used in a V.32 modem, is shown in Figure 2. A detailed illustration of the actual filter used in the baseband echo canceler can be found in Figure 3. Note that the single lines indicate real signals, while...

16/3,K/3 (Item 1 from file: 88)
DIALOG(R)File 88:Gale Group Business A.R.T.S.
(c) 2003 The Gale Group. All rts. reserv.

04077338 SUPPLIER NUMBER: 18830886

Optimum filter banks for signal decomposition and its application in adaptive echo cancellation.

Jin, Qu; Luo, Zhi-Quan; Wong, Kon Max

IEEE Transactions on Signal Processing, v44, n7, p1669(12)

July, 1996

ISSN: 1053-587X LANGUAGE: English RECORD TYPE: Abstract

ABSTRACT: The optimum quasi-biorthogonal (OQB) filter banks allow efficient echo cancellation in electronic signals. A multiresolution algorithm decomposes a distorted original signal into a number of

June 27, 2003

components. An adaptive algorithm **cancels** the **echo** in the received signal. The filters have high energy concentration in the **passband**. The use of adjacent-band **adaptive filtering** along with OQB filters gives better performance than that obtained using in-band filtering alone.

16/3,K/4 (Item 2 from file: 88)
DIALOG(R)File 88:Gale Group Business A.R.T.S.
(c) 2003 The Gale Group. All rts. reserv.

03447686 SUPPLIER NUMBER: 15061353
Neural networks: applications in industry, business and science.
(Artificial Intelligence) (Cover Story) (Technical)
Widrow, Bernard; Rumelhart, David E.; Lehr, Michael A.
Communications of the ACM, v37, n3, p93(13)
March, 1994
DOCUMENT TYPE: Technical ISSN: 0001-0782 LANGUAGE: English
RECORD TYPE: Fulltext; Abstract
WORD COUNT: 7127 LINE COUNT: 00713

... lines, which would normally be tolerated with speech, is devastating to high-speed data transmission. **Echo cancelling** solves the problem by detecting the echo and adding an equal and opposite signal to the return path. The cancelling signal is generated by an **adaptive transversal filter** whose coefficients (weights) are automatically adjusted by the LMS algorithm of Widrow and Hoff [32], also known as the delta rule in the field of neural networks. The **adaptive filter** makes use of what amounts to a single neuron. The first **echo cancellers** were developed at AT&T Bell Labs in the 1960s by M. M. Sondhi and...

...fiber-optic channels can have nonflat frequency responses and nonlinear phase responses in the signal **passband**. Sending digital data at high speed through these channels often results in a phenomenon called...

...medium. Equalization in data modems combats this phenomenon by filtering incoming signals. A modem's **adaptive filter**, by **adapting** itself to become a channel inverse, can compensate for the irregularities in channel magnitude and...

16/3,K/5 (Item 1 from file: 275)
DIALOG(R)File 275:Gale Group Computer DB(TM)
(c) 2003 The Gale Group. All rts. reserv.

01674631 SUPPLIER NUMBER: 15061353 (USE FORMAT 7 OR 9 FOR FULL TEXT)
Neural networks: applications in industry, business and science.
(Artificial Intelligence) (Cover Story) (Technical)
Widrow, Bernard; Rumelhart, David E.; Lehr, Michael A.
Communications of the ACM, v37, n3, p93(13)
March, 1994
DOCUMENT TYPE: Technical ISSN: 0001-0782 LANGUAGE: ENGLISH
RECORD TYPE: FULLTEXT; ABSTRACT
WORD COUNT: 8599 LINE COUNT: 00713

... lines, which would normally be tolerated with speech, is devastating to high-speed data transmission. **Echo cancelling** solves the problem by detecting the echo and adding an equal and opposite signal to the return path. The cancelling signal is generated by an **adaptive transversal filter** whose coefficients (weights) are automatically adjusted by the LMS algorithm of Widrow and Hoff [32], also known as the delta rule in the field of neural networks. The **adaptive filter** makes use of what amounts to a single neuron. The first **echo cancellers** were developed at AT&T Bell Labs in the 1960s by M. M. Sondhi and...

...fiber-optic channels can have nonflat frequency responses and nonlinear

June 27, 2003

phase responses in the signal **passband**. Sending digital data at high speed through these channels often results in a phenomenon called...

...medium. Equalization in data modems combats this phenomenon by filtering incoming signals. A modem's **adaptive filter**, by adapting itself to become a channel inverse, can compensate for the irregularities in channel magnitude and...

16/3,K/6 (Item 1 from file: 15)
DIALOG(R)File 15:ABI/Inform(R)
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00389602 88-06435
Echo Cancellation for High-Speed Dial-Up Applications: Part I
Turner, Steven E.
Telecommunications v22n1 (North American Edition) PP: 80-87, 104 Jan 1988
ISSN: 0040-2494 JRNL CODE: TEC

ABSTRACT: **Echo canceler** technology has arrived with the advent of the V.32 modem. A critical design issue in the field of data communication, **echo canceler** technology must be understood by design engineers and technical managers. Basically, an **echo canceler** is a filter used to model the echo path created when a transmitted signal flows through the telephone network. Types of **echo cancelers** include data-domain **echo cancelers**, which can be either baseband or **passband** cancelers or a combination of both. Differences between baseband and **passband** cancelers are: 1. the type of signal contained in the **adaptive FIR canceler filter**, and 2. the location and technique by which the actual cancellation takes place. While the baseband canceler requires significant use of complex arithmetic, the **passband** canceler takes much longer to converge and train. The Weinstein canceler is the combined version...
... a short baseband modeling filter while avoiding the complex arithmetic at the point of actual **echo cancellation**.

16/3,K/7 (Item 1 from file: 647)
DIALOG(R)File 647:CMP Computer Fulltext
(c) 2003 CMP Media, LLC. All rts. reserv.

00508707 CMP ACCESSION NUMBER: EET19921012S1572
FOUR-CHIP SET COMBINES FAX, DATA, VOICE:Sierra modem chip set
LORING WIRBEL
ELECTRONIC ENGINEERING TIMES, 1992, n 714, 21
PUBLICATION DATE: 921012
JOURNAL CODE: EET LANGUAGE: English
RECORD TYPE: Fulltext
SECTION HEADING: News: Business
WORD COUNT: 567

... analog front end includes 12-bit A/D and D/A converters, a seventh-order low - pass receive **filter**, and **adaptive** features for near-end **echo cancellation**.

The SC11083 interface IC sweeps up glue logic for the AT bus, parallel to serial...

June 27, 2003

File 348:EUROPEAN PATENTS 1978-2003/Jun W04

(c) 2003 European Patent Office

File 349:PCT FULLTEXT 1979-2002/UB=20030626, UT=20030619

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Set	Items	Description
S1	6	AU='GAO SHAWN':AU='GAO SHAWN X'
S2	13	AU='SOLI SIGFRID':AU='SOLI SIGFRIED D'
S3	4	AU='GHI'
S4	0	S1 AND S2 AND S3
S5	4	S1 AND S2
S6	0	S1 AND S3

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5/5,K/1 (Item 1 from file: 348)
DIALOG(R) File 348:EUROPEAN PATENTS
(c) 2003 European Patent Office. All rts. reserv.

01154395

BAND-LIMITED ADAPTIVE FEEDBACK CANCELLER FOR HEARING AIDS
BANDBEGRENZTE ADAPTIVE RUCKKOPPLUNGSUNTERDRUCKUNG FUR HORHILFEGERATE
DISPOSITIF ADAPTATIF DE SUPPRESSION DE L'EFFET LARSEN A BANDE LIMITEE
DESTINE AUX PROTHESES AUDITIVES

PATENT ASSIGNEE:

HOUSE EAR INSTITUTE, (2139940), 5th floor, 21000 West Thirt Street, Los Angeles, CA 90057, (US), (Applicant designated States: all)

INVENTOR:

GAO, Shawn , 18304 Susan Place, Cerritos, CA 91024, (US)
SOLI, Sigfrid , 2020 North Santa Anita Avenue, Sierra Madre, CA 91024, (US)

CHI, Hsiang-Feng, 16907 Larbrook Drive, Hacienda Heights, CA 91745, (US)

LEGAL REPRESENTATIVE:

Wombwell, Francis (46021), Potts, Kerr & Co. 15, Hamilton Square, Birkenhead Merseyside L41 6BR, (GB)

PATENT (CC, No, Kind, Date): EP 1118247 A2 010725 (Basic)
WO 200019605 000406

APPLICATION (CC, No, Date): EP 99948516 990930; WO 99US22757 990930

PRIORITY (CC, No, Date): US 102557 P 980930

DESIGNATED STATES: AT; BE; CH; CY; DE; DK; ES; FI; FR; GB; GR; IE; IT; LI; LU; MC; NL; PT; SE

EXTENDED DESIGNATED STATES: AL; LT; LV; MK; RO; SI

INTERNATIONAL PATENT CLASS: H04R-025/00; H04R-003/02

NOTE:

No A-document published by EPO

LEGAL STATUS (Type, Pub Date, Kind, Text):

Application: 000531 A2 International application. (Art. 158(1))

Application: 000531 A2 International application entering European phase

Application: 010725 A2 Published application without search report

Examination: 010725 A2 Date of request for examination: 20010514

LANGUAGE (Publication, Procedural, Application): English; English; English

INVENTOR:

GAO, Shawn ...

...US)

SOLI, Sigfrid ...

5/5,K/2 (Item 2 from file: 348)

DIALOG(R) File 348:EUROPEAN PATENTS
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01037051

METHOD OF MEASURING AND PREVENTING UNSTABLE FEEDBACK IN HEARING AIDS
PROCEDE DE MESURE ET DE PREVENTION DE RETROACTION INSTABLE DANS DES PROTHESES AUDITIVES

PATENT ASSIGNEE:

HOUSE EAR INSTITUTE, (2139940), 5th floor, 21000 West Thirt Street, Los Angeles, CA 90057, (US), (Applicant designated States: all)

INVENTOR:

GAO, Shawn, X. , 5th floor 2100 West Third Street, Los Angeles, CA 90057, (US)

SOLI, Sigfrid, D. , 5th floor 2100 West Third Street, Los Angeles, CA 90057, (US)

PATENT (CC, No, Kind, Date):

WO 9912388 990311

APPLICATION (CC, No, Date): WO 98946842 980903; WO 98US18442 980903

PRIORITY (CC, No, Date): US 926320 970905

June 27, 2003

DESIGNATED STATES: AT; BE; CH; CY; DE; DK; ES; FI; FR; GB; GR; IE; IT; LI;
LU; MC; NL; PT; SE

INTERNATIONAL PATENT CLASS: H04R-025/00

LEGAL STATUS (Type, Pub Date, Kind, Text):

Application: 010131 A1 International application. (Art. 158(1))

Application: 990602 A1 International application (Art. 158(1))

Withdrawal: 010131 A1 Date application deemed withdrawn: 20000406

Appl Changed: 010131 A1 International application not entering European
phase

LANGUAGE (Publication, Procedural, Application): English; English; English

INVENTOR:

GAO, Shawn, X ...

...US)

SOLI, Sigfrid, D ...

5/5,K/3 (Item 1 from file: 349)

DIALOG(R) File 349:PCT FULLTEXT

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00556232 **Image available**

BAND-LIMITED ADAPTIVE FEEDBACK CANCELLER FOR HEARING AIDS

DISPOSITIF ADAPTATIF DE SUPPRESSION DE L'EFFET LARSEN A BANDE LIMITEE
DESTINE AUX PROTHESES AUDITIVES

Patent Applicant/Assignee:

HOUSE EAR INSTITUTE,

Inventor(s):

GAO Shawn ,

SOLI Sigfrid ,

CHI Hsiang-Feng

Patent and Priority Information (Country, Number, Date):

Patent: WO 200019605 A2 20000406 (WO 0019605)

Application: WO 99US22757 19990930 (PCT/WO US9922757)

Priority Application: US 98102557 19980930

Designated States: AE AL AM AT AU AZ BA BB BG BR BY CA CH CN CR CU CZ CZ DE
DE DK DM EE EE ES FI FI GB GD GE GH GM HR HU ID IL IN IS JP KE KG KP
KR KZ LC LK LR LS LT LU LV MD MG MK MN MW MX NO NZ PL PT RO RU SD SE SG
SI SK SL TJ TM TR TT TZ UA UG UZ VN YU ZA ZW GH GM KE LS MW SD SL SZ
TZ ÜG ZW AM AZ BY KG KZ MD RU TJ TM AT BE CH CY DE DK ES FI FR GB GR IE
IT LU MC NL PT SE BF BJ CF CG CI CM GA GN GW ML MR NE SN TD TG

Main International Patent Class: H04R-025/00

International Patent Class: H04R-003/02

Publication Language: English

Fulltext Availability:

Detailed Description

Claims

Fulltext Word Count: 9255

English Abstract

An improved method for adaptively cancelling acoustic feedback in hearing aids and other audio amplification devices. Feedback cancellation is limited to a frequency band that encompasses all unstable frequencies. By limiting the bandwidth of the feedback cancellation signal, the distortion due to the adaptive filter is minimized and limited only to the unstable feedback regions. A relatively simple signal processing algorithm is used to produce highly effective results with minimal signal distortion.

French Abstract

L'invention concerne un procede ameliore pour supprimer de maniere adaptative l'effet Larsen dans les protheses auditives et dans d'autres dispositifs audio amplifies. La suppression de l'effet Larsen est limitee a la bande de frequences qui englobe toutes les frequences instables. En

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limitant la bande de fréquences du signal d'annulation de l'effet Larsen, on arrive à réduire au minimum la distorsion provoquée par le filtre adaptatif, qui est limitée uniquement aux zones instables de l'effet Larsen. On utilise un algorithme relativement simple de traitement des signaux pour obtenir des résultats probants, et ce avec une distorsion minimale des signaux.

Inventor(s):
GAO Shawn ...
... SOLI Sigfrid

5/5,K/4 (Item 2 from file: 349)
DIALOG(R) File 349:PCT FULLTEXT
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00481036 **Image available**
METHOD OF MEASURING AND PREVENTING UNSTABLE FEEDBACK IN HEARING AIDS
PROCEDE DE MESURE ET DE PREVENTION DE RETROACTION INSTABLE DANS DES
PROTHESES AUDITIVES

Patent Applicant/Assignee:
HOUSE EAR INSTITUTE,

Inventor(s):

GAO Shawn X ,
SOLI Sigfrid D

Patent and Priority Information (Country, Number, Date):

Patent: WO 9912388 A1 19990311
Application: WO 98US18442 19980903 (PCT/WO US9818442)
Priority Application: US 97926320 19970905

Designated States: AL AM AT AT AU AZ BA BB BG BR BY CA CH CN CU CZ CZ DE DE
DK DK EE EE ES FI FI GB GE GH GM HR HU ID IL IS JP KE KG KP KR KZ LC LK
LR LS LT LU LV MD MG MK MN MW MX NO NZ PL PT RO RU SD SE SG SI SK SK SL
TJ TM TR TT UA UG UZ VN YU ZW GH GM KE LS MW SD SZ UG ZW AM AZ BY KG KZ
MD RU TJ TM AT BE CH CY DE DK ES FI FR GB GR IE IT LU MC NL PT SE BF BJ
CF CG CI CM GA GN GW ML MR NE SN TD TG

Main International Patent Class: H04R-025/00

Publication Language: English

Fulltext Availability:

Detailed Description
Claims

Fulltext Word Count: 6962

English Abstract

A "true" hearing aid transfer function ($K(f)$), including feedback, is derived from measurements taken with the hearing aid fitted in a patient's ear canal. Closed loop transfer functions ($L(f)$) are calculated at several hearing aid gains without opening the internal circuitry of the hearing aid using a time domain Weiner optimal filter model. The combined open loop transfer function of the hearing aid and feedback path is then calculated. Once the open loop transfer function is known, potentially unstable frequencies are identified and maximum hearing aid gain settings are determined. The hearing aid transfer function and transfer function of feedback path ($B(f)$) are also calculated from the closed loop transfer function measurements.

French Abstract

L'invention concerne une fonction "véritable" de transfert de prothèse auditive ($K(f)$), y compris une rétroaction, dérivée de mesures prises avec la prothèse auditive ajustée dans le conduit auditif externe d'un patient. On calcule des fonctions de transfert en boucle fermée ($L(f)$) à plusieurs gains de prothèse auditive sans ouvrir l'ensemble de circuits internes de la prothèse auditive grâce à un modèle de filtre optimal de Weiner à réponse temporelle. On calcule ainsi la fonction de transfert en boucle ouverte combinée de la prothèse auditive et de la trajectoire de

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retroaction. Une fois que la fonction de transfert en boucle ouverte est connue, on identifie les frequences potentiellement instables et on determine les reglages de gains de prothese auditive maximaux. On calcule egalement la fonction de transfert de la prothese auditive et la fonction de transfert de la trajectoire de retroaction ($B(f)$), a partir des mesures de fonction de transfert en boucle fermee.

Inventor(s):

GAO Shawn X ...

... SOLI Sigfrid D

June 27, 2003

File 344:Chinese Patents Abs Aug 1985-2003/Mar
(c) 2003 European Patent Office
File 347:JAPIO Oct 1976-2003/Feb(Updated 030603)
(c) 2003 JPO & JAPIO
File 350:Derwent WPIX 1963-2003/UD,UM &UP=200340
(c) 2003 Thomson Derwent

Set	Items	Description
S1	200	AU='GAO S'
S2	11	AU='SOLI S':AU='SOLI S D'
S3	0	AU='GHI H'
S4	2	S1 AND S2

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4/5/1 (Item 1 from file: 350)

DIALOG(R) File 350:Derwent WPIX

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013145713 **Image available**

WPI Acc No: 2000-317585/200027

XRPX Acc No: N00-238393

Adaptive feedback canceller for audio amplification devices e.g. hearing aid, has band limited filter with passband encompassing all unstable frequencies

Patent Assignee: HOUSE EAR INST (HOUS-N)

Inventor: CHI H; GAO S; SOLI S

Number of Countries: 089 Number of Patents: 004

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
WO 200019605	A2	20000406	WO 99US22757	A	19990930	200027 B
AU 9961680	A	20000417	AU 9961680	A	19990930	200035
EP 1118247	A2	20010725	EP 99948516	A	19990930	200143
			WO 99US22757	A	19990930	
JP 2002526961	W	20020820	WO 99US22757	A	19990930	200258
			JP 2000572997	A	19990930	

Priority Applications (No Type Date): US 98102557 P 19980930

Patent Details:

Patent No Kind Lan Pg Main IPC Filing Notes

WO 200019605 A2 E 66 H03H-021/00

Designated States (National): AE AL AM AT AU AZ BA BB BG BR BY CA CH CN CR CU CZ DE DK DM EE ES FI GB GD GE GH GM HR HU ID IL IN IS JP KE KG KP KR KZ LC LK LR LS LT LU LV MD MG MK MN MW MX NO NZ PL PT RO RU SD SE SG SI SK SL TJ TM TR TT TZ UA UG UZ VN YU ZA ZW

Designated States (Regional): AT BE CH CY DE DK EA ES FI FR GB GH GM GR IE IT KE LS LU MC MW NL OA PT SD SE SL SZ TZ UG ZW

AU 9961680 A H03H-021/00 Based on patent WO 200019605

EP 1118247 A2 E H04R-025/00 Based on patent WO 200019605

Designated States (Regional): AL AT BE CH CY DE DK ES FI FR GB GR IE IT LI LT LU LV MC MK NL PT RO SE SI

JP 2002526961 W 58 H04B-003/23 Based on patent WO 200019605

Abstract (Basic): WO 200019605 A2

NOVELTY - The feedback canceller includes an adaptive digital filter (30) whose output is combined with input of audio amplification device. A band limiting filter having passband limited to a frequency band containing unstable frequencies is coupled between the amplification device and adaptive filter.

DETAILED DESCRIPTION - An INDEPENDENT CLAIM is also included for method for adaptively canceling acoustic feedback.

USE - For audio amplification devices such as hearing aid.

ADVANTAGE - By limiting the bandwidth of the feedback cancellation signal, distortion due to adaptive filter is minimized and limited only to unstable feedback regions.

DESCRIPTION OF DRAWING(S) - The figure shows functional block diagram of hearing aid and feedback canceller.

Adaptive digital filter (30)

PP; 66 DwgNo 4/21

Title Terms: ADAPT; FEEDBACK; CANCEL; AUDIO; AMPLIFY; DEVICE; HEARING; AID; BAND; LIMIT; FILTER; PASSBAND; ENCOMPASSING; UNSTABLE; FREQUENCY

Derwent Class: U22; U25

International Patent Class (Main): H04B-003/23; H04R-025/00

International Patent Class (Additional): H03H-021/00; H04R-003/02

File Segment: EPI

4/5/2 (Item 2 from file: 350)

DIALOG(R) File 350:Derwent WPIX

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009941285 **Image available**

WPI Acc No: 1994-208997/199425

XRPX Acc No: N94-164458

Signal processing to maintain directional hearing with hearing aid -
using filter compensating for insertion effect derived from ratio of
unaided to aided head related transfer function

Patent Assignee: HOUSE EAR INST (HOUS-N)

Inventor: GAO S ; JAYARAMAN S; SOLI S D ; SULLIVAN J

Number of Countries: 001 Number of Patents: 001

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
US 5325436	A	19940628	US 9385652	A	19930630	199425 B

Priority Applications (No Type Date): US 9385652 A 19930630

Patent Details:

Patent No	Kind	Lan Pg	Main IPC	Filing Notes
US 5325436	A	17	H04R-005/00	

Abstract (Basic): US 5325436 A

The method for obtaining coefficients of a digital filter for use in compensating effects of a hearing aid involves determining an unaided head related transfer function for each ear and for several azimuth locations of a sound source. An aided head related transfer function is determined for each ear using a hearing aid for the several azimuth locations of the sound source.

Minimum phase representation of the unaided and aided head related transfer function are found. The ratio between the unaided and aided minimum phase representation is calculated to form a target filter response. Several filter coefficients are obtained by sampling the target filter response at several frequency values corresponding to frequency increments in the digital filter.

USE/ADVANTAGE - Allows user to determine direction of sound.

Dwg.2/10

Title Terms: SIGNAL; PROCESS; MAINTAIN; DIRECTION; HEARING; HEARING; AID; FILTER; COMPENSATE; INSERT; EFFECT; DERIVATIVE; RATIO; UNAIDED; AID; HEAD ; RELATED; TRANSFER; FUNCTION

Derwent Class: U22; W04

International Patent Class (Main): H04R-005/00

International Patent Class (Additional): H04R-025/00; H04R-029/00

File Segment: EPI

June 27, 2003

File 344:Chinese Patents Abs Aug 1985-2003/Mar
(c) 2003 European Patent Office
File 347:JAPIO Oct 1976-2003/Feb(Updated 030603)
(c) 2003 JPO & JAPIO
File 350:Derwent WPIX 1963-2003/UD,UM &UP=200340
(c) 2003 Thomson Derwent

Set	Items	Description
S1	90144	(AUDIO? OR AMPLIFICATION? OR ACOUSTIC? OR HEARING) (2N) (AID? OR DEVICE? OR APPARATUS? OR APPTS) OR SPEAKER?
S2	622371	FILTER?
S3	83703	(BAND? OR HIGH? OR LOW?) () (PASS OR STOP? OR LIMIT?) OR PAS- SBAND?
S4	16615	(FEEDBACK? OR ECHO?) (3N) (LOOP? OR CANCEL? OR PREVENT?)
S5	33	S1 AND S2 AND S3 AND S4
S6	158	S1(3N)S4
S7	6	S6 AND S2 AND S3
S8	329	S2()ADAPT?
S9	0	S8 AND S1 AND S3 AND S4
S10	148	S3(5N)S4
S11	7	S10 AND S1 AND S2
S12	6	S11 NOT S7
S13	21	S5 NOT (S7 OR S12)
S14	6	S13 AND IC=H04R-025/00
S15	15	S13 NOT S14

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7/5/1 (Item 1 from file: 347)
DIALOG(R) File 347:JAPIO
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02677118 **Image available**
LOUD-SPEAKER TELEPHONE SET

PUB. NO.: 63-294018 [JP 63294018 A]
PUBLISHED: November 30, 1988 (19881130)
INVENTOR(s): ITO YOSHIO
MIYAMOTO RYOICHI
NAKANO YOSHIKAZU
APPLICANT(s): OKI ELECTRIC IND CO LTD [000029] (A Japanese Company or Corporation), JP (Japan)
APPL. NO.: 62-127935 [JP 87127935]
FILED: May 27, 1987 (19870527)
INTL CLASS: [4] H04B-003/23; H04M-001/60; H04M-009/08
JAPIO CLASS: 44.2 (COMMUNICATION -- Transmission Systems); 44.4 (COMMUNICATION -- Telephone)
JOURNAL: Section: E, Section No. 734, Vol. 13, No. 126, Pg. 85, March 28, 1989 (19890328)

ABSTRACT

PURPOSE: To obtain a loud-speaker telephone set for automobile superior in service quality by erasing a passing-round signal only in a certain partial band of the voice signal band for the purpose of preventing the signal, which a speaker in the automobile sends, from passing round to the speaker after a certain time.

CONSTITUTION: A transmission signal S_k from a microphone 3 is inputted to a low - pass filter 14, and only its low band component SL_k is taken out and is inputted to an echo canceller 15. An estimated value $-rL_k$ of a low band component rL_k from a band separating filter 16 is estimated in the canceller 15, and its phase inverted signal is inputted to an adder 17. Consequently, the signal rL_k is erased in the output of the adder 17. Its erase error eL_k is inputted to a band synthesizing filter 18. A high band component rH_k from the filter 16 and the erase error eL_k are synthesized into an original voice band signal RD_k by the filter 18. This signal is inputted to a speaker 2 and an echo canceller 1. Consequently, the low band component is erased from a passing-round signal r_k of the signal inputted from the microphone 3 with respect to the signal RD_k .

7/5/2 (Item 1 from file: 350)
DIALOG(R) File 350:Derwent WPIX
(c) 2003 Thomson Derwent. All rts. reserv.

013145713 **Image available**
WPI Acc No: 2000-317585/200027
XRPX Acc No: N00-238393

Adaptive feedback canceller for audio amplification devices
e.g. hearing aid, has band limited filter with passband
encompassing all unstable frequencies

Patent Assignee: HOUSE EAR INST (HOUS-N)

Inventor: CHI H; GAO S; SOLI S

Number of Countries: 089 Number of Patents: 004

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
WO 200019605	A2	20000406	WO 99US22757	A	19990930	200027 B
AU 9961680	A	20000417	AU 9961680	A	19990930	200035
EP 1118247	A2	20010725	EP 99948516	A	19990930	200143
			WO 99US22757	A	19990930	
JP 2002526961	W	20020820	WO 99US22757	A	19990930	200258
			JP 2000572997	A	19990930	

June 27, 2003

Priority Applications (No Type Date): US 98102557 P 19980930

Patent Details:

Patent No Kind Lan Pg Main IPC Filing Notes

WO 200019605 A2 E 66 H03H-021/00

Designated States (National): AE AL AM AT AU AZ BA BB BG BR BY CA CH CN CR CU CZ DE DK DM EE ES FI GB GD GE GH GM HR HU ID IL IN IS JP KE KG KP KR KZ LC LK LR LS LT LU LV MD MG MK MN MW MX NO NZ PL PT RO RU SD SE SG SI SK SL TJ TM TR TT TZ UA UG UZ VN YU ZA ZW

Designated States (Regional): AT BE CH CY DE DK EA ES FI FR GB GH GM GR IE IT KE LS LU MC MW NL OA PT SD SE SL SZ TZ UG ZW

AU 9961680 A H03H-021/00 Based on patent WO 200019605

EP 1118247 A2 E H04R-025/00 Based on patent WO 200019605

Designated States (Regional): AL AT BE CH CY DE DK ES FI FR GB GR IE IT LI LT LU LV MC MK NL PT RO SE SI

JP 2002526961 W 58 H04B-003/23 Based on patent WO 200019605

Abstract (Basic): WO 200019605 A2

NOVELTY - The feedback canceller includes an adaptive digital filter (30) whose output is combined with input of audio amplification device. A band limiting filter having passband limited to a frequency band containing unstable frequencies is coupled between the amplification device and adaptive filter.

DETAILED DESCRIPTION - An INDEPENDENT CLAIM is also included for method for adaptively canceling acoustic feedback.

USE - For audio amplification devices such as hearing aid.

ADVANTAGE - By limiting the bandwidth of the feedback cancellation signal, distortion due to adaptive filter is minimized and limited only to unstable feedback regions.

DESCRIPTION OF DRAWING(S) - The figure shows functional block diagram of hearing aid and feedback canceller.

Adaptive digital filter (30)

pp; 66 DwgNo 4/21

Title Terms: ADAPT; FEEDBACK; CANCEL; AUDIO; AMPLIFY; DEVICE; HEARING; AID; BAND; LIMIT; FILTER ; PASSBAND ; ENCOMPASSING; UNSTABLE; FREQUENCY

Derwent Class: U22; U25

International Patent Class (Main): H04B-003/23; H04R-025/00

International Patent Class (Additional): H03H-021/00; H04R-003/02

File Segment: EPI

7/5/3 (Item 2 from file: 350)

DIALOG(R) File 350:Derwent WPIX

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012858799 **Image available**

WPI Acc No: 2000-030632/200003

XRPX Acc No: N00-023656

Amplification type intercom apparatus - includes microphones in input side of echo canceler and speakers connected in output side of echo cancelers in both base- station and sub-station

Patent Assignee: AIHON KK (AIHO-N)

Number of Countries: 001 Number of Patents: 001

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
JP 11298618	A	19991029	JP 9899263	A	19980410	200003 B

Priority Applications (No Type Date): JP 9899263 A 19980410

Patent Details:

Patent No Kind Lan Pg Main IPC Filing Notes

JP 11298618 A 12 H04M-009/08

Abstract (Basic): JP 11298618 A

NOVELTY - Echo cancelers (M15,T15) of base-station (M1) and sub-station (T1) are connected to the output side of high pass filters (M22,T22), respectively. Microphones (M12,T12) are connected

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to input of each **echo canceler**, and **speakers** (M17,T17) are connected to the output of echo cancelers. **High pass filters** (M22,T22) are connected to input of codec (M16,T16) connected to the line (L1).

USE - For e.g. hands-free speaker phone.

ADVANTAGE - Enables transmission and reception of more audio signals having high frequency. DESCRIPTION OF DRAWING(S) - The figure shows the block diagram of intercom apparatus. (L1) Line; (M1) Base-station; (M12,T12) Microphones; (M15,T15) Echo cancelers; (M16,T16) Codec; (M17,T17) Speakers; (M22,T22) **High pass filters**; (T1) Sub-station.

Dwg.1/3

Title Terms: AMPLIFY; TYPE; INTERCOMMUNICATION; APPARATUS; MICROPHONE; INPUT; SIDE; ECHO; SPEAKER; CONNECT; OUTPUT; SIDE; ECHO; BASE; STATION; SUB; STATION

Derwent Class: W01

International Patent Class (Main): H04M-009/08

International Patent Class (Additional): H04M-001/60

File Segment: EPI

7/5/4 (Item 3 from file: 350)

DIALOG(R) File 350:Derwent WPIX

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010974026 **Image available**

WPI Acc No: 1996-470975/199647

XRPX Acc No: N96-397198

Howling canceler for preventing feedback of sound from speaker to microphone - has digital-analog converter which performs digital-analog conversion of each frequency-band signal after concerned signal has been controlled to stable side

Patent Assignee: JAPAN RADIO CO LTD (NIUR)

Number of Countries: 001 Number of Patents: 001

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
JP 8237789	A	19960913	JP 9558263	A	19950223	199647 B

Priority Applications (No Type Date): JP 9558263 A 19950223

Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
JP 8237789	A	3		H04R-003/02	

Abstract (Basic): JP 8237789 A

The canceler includes an analog-digital converter (2) which converts an audio signal input to a microphone to a digital signal. The obtd. digital signal is divided into several frequency-band signal of desired number of channels through a **band - pass filter**. An oscillation sensor detects the existence of oscillation at each frequency-band signal.

An adder (6) combines the frequency band signals that passes to an oscillation prevention circuit. A digital-analog converter (7) performs the digital-analog conversion of each frequency-band signal after the concerned signal has been controlled to a stable side.

ADVANTAGE - Offers howling canceler which pertinently prevents howling during signal processing. Prevents changing of microphone direction and specific speaker interruption.

Dwg.1/2

Title Terms: HOWLING; PREVENT; FEEDBACK; SOUND; SPEAKER; MICROPHONE; DIGITAL; ANALOGUE; CONVERTER; PERFORMANCE; DIGITAL; ANALOGUE; CONVERT; FREQUENCY; BAND; SIGNAL; AFTER; CONCERN; SIGNAL; CONTROL; STABILISED; SIDE

Derwent Class: U25; W04

International Patent Class (Main): H04R-003/02

International Patent Class (Additional): H03G-005/14

June 27, 2003

File Segment: EPI

7/5/5 (Item 4 from file: 350)

DIALOG(R) File 350:Derwent WPIX
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010463052 **Image available**

WPI Acc No: 1995-364371/199547

XRPX Acc No: N95-269538

Loud-speaker terminal equipment - has alarm tone transmission function
corresp. to volume approved by fire-fighting authority with speaker
built with acoustic feedback loop between microphones

Patent Assignee: AIHON KK (AIHO-N)

Number of Countries: 001 Number of Patents: 001

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
JP 7248789	A	19950926	JP 9442489	A	19940314	199547 B

Priority Applications (No Type Date): JP 9442489 A 19940314

Patent Details:

Patent No	Kind	Lan Pg	Main IPC	Filing Notes
JP 7248789	A	4	G10K-015/04	

Abstract (Basic): JP 7248789 A

The equipment includes a microphone (1) that generates a specific frequency signal (f2) from an audio signal (f1) that passes through a band - pass filter (3).

An electronic switch (4) is activated according to an alarm signal (f6) produced by a sensor (19) in passing the specific frequency signal into a speaker (10) which is built with an acoustic-feedback loop between the microphones.

ADVANTAGE - Enables reduction of electric audio conversion efficiency caused by change in manufacturing error of speaker without enlarging power amplifier that drives it.

Dwg.1/3

Title Terms: LOUD; SPEAKER; TERMINAL; EQUIPMENT; ALARM; TONE; TRANSMISSION; FUNCTION; CORRESPOND; VOLUME; APPROVE; FIRE; FIGHTING; AUTHORISE; SPEAKER ; BUILD; ACOUSTIC; FEEDBACK; LOOP; MICROPHONE

Derwent Class: P86; W01; W05

International Patent Class (Main): G10K-015/04

International Patent Class (Additional): H04M-011/04; H04R-003/04

File Segment: EPI; EngPI

7/5/6 (Item 5 from file: 350)

DIALOG(R) File 350:Derwent WPIX

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007529265 **Image available**

WPI Acc No: 1988-163197/198824

XRPX Acc No: N88-124686

Super-regenerative detector with saw device e.g. for portable phone - has quench oscillator to switch RF oscillator including SAW device between oscillation and non-oscillation

Patent Assignee: RF MONOLITHICS INC (RFMO-N)

Inventor: ASH D L

Number of Countries: 007 Number of Patents: 005

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
EP 271190	A	19880615	EP 87308927	A	19871008	198824 B
US 4749964	A	19880607	US 86939527	A	19861208	198825
JP 63198404	A	19880817	JP 87304646	A	19871203	198839
EP 271190	B1	19940302	EP 87308927	A	19871008	199409
DE 3789206	G	19940407	DE 3789206	A	19871008	199415

June 27, 2003

EP 87308927 A 19871008

Priority Applications (No Type Date): US 86939527 A 19861208
Cited Patents: 1.Jnl.Ref; A3...8922; EP 184508; FR 2209255; No-SR.Pub; US
3119065; US 3405364; US 4143324

Patent Details:

Patent No	Kind	Lan Pg	Main IPC	Filing Notes
EP 271190	A	E 39		
Designated States (Regional): DE FR GB IT NL				
US 4749964	A	8		
EP 271190	B1	E 9	H03D-011/04	
Designated States (Regional): DE FR GB IT NL				
DE 3789206	G		H03D-011/04	Based on patent EP 271190

Abstract (Basic): EP 271190 A

The detector circuit uses a single transistor (Q1) to the collector of which the modulated RF signal is coupled from a terminal terminal (20) through a coupling capacitor (C5). The RF signal is coupled also through a **feedback loop** including the surface **acoustic wave device** (22) and an inductor (L2) to initiate oscillations more rapidly than is the case with thermal noise alone as the input voltage.

The SAW device has a relatively low quality factor and low loss. The transistor output is coupled through an inductor (L1) and capacitor (C3) to a **low pass filter** (24) for recovery of the modulation signal.

USE/ADVANTAGE - In garage door opening receiver. Is temperature stable, does not drift in frequency, and has very narrow reception band eliminating effects of noise and stray signals

Title Terms: SUPER; REGENERATE; DETECT; SAW; DEVICE; PORTABLE; TELEPHONE; QUENCH; OSCILLATOR; SWITCH; RF; OSCILLATOR; SAW; DEVICE; OSCILLATING; NON ; OSCILLATING

Index Terms/Additional Words: SUPER; REGENERATE; DETECT; SAW; DEVICE; PORTABLE

Derwent Class: U14; U23; W01; W02; W05; X25

International Patent Class (Main): H03D-011/04

International Patent Class (Additional): H03B-005/00; H03D-001/18

File Segment: EPI

June 27, 2003

12/5/1 (Item 1 from file: 347)
DIALOG(R) File 347:JAPIO
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06357010 **Image available**
LOUD SPEAKER INTERPHONE SYSTEM

PUB. NO.: 11-298618 [JP 11298618 A]
PUBLISHED: October 29, 1999 (19991029)
INVENTOR(s): NISHIMURA TOMOHIRO
KITAGAWA KAZUMI
APPLICANT(s): AIPHONE CO LTD
APPL. NO.: 10-099263 [JP 9899263]
FILED: April 10, 1998 (19980410)
INTL CLASS: H04M-009/08; H04M-001/60

ABSTRACT

PROBLEM TO BE SOLVED: To transmit/receive audio signals at a much higher frequency without being limited to audio signals in a telephone band limited by an echo canceler with respect to a loud speaker interphone system.

SOLUTION: Among audio signals exchanged between a base unit M1 and a hand set T1 connected through a line L1, audio signals in the telephone band are respectively discharged from first speakers M11 and T11 through first and third band-pass filters M21 T21, M23 and T23 and echo cancelers M15 and T15 and audio signals at the frequency higher than the telephone band are respectively discharged from second speakers M17 and T17 separately provided under the control of audio switches M19 and T19 through second and fourth band-pass switches M22, T22, M24 and T24. Thus, the audio signals in the telephone band can be made into fully duplex communication and the audio signals at the frequency higher than the telephone band can be made into semi-duplex communication.

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12/5/2 (Item 2 from file: 347)
DIALOG(R) File 347:JAPIO
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05449992 **Image available**
LOUD SPEAKER INFORMATION COMMUNICATION SYSTEM

PUB. NO.: 09-064792 [JP 9064792 A]
PUBLISHED: March 07, 1997 (19970307)
INVENTOR(s): IWASAKI TAKASHI
KUSANO YOSHIMASA
APPLICANT(s): KYOCERA CORP [358923] (A Japanese Company or Corporation), JP (Japan)
APPL. NO.: 07-210495 [JP 95210495]
FILED: August 18, 1995 (19950818)
INTL CLASS: [6] H04B-003/23; H03H-021/00; H04M-001/60
JAPIO CLASS: 44.2 (COMMUNICATION -- Transmission Systems); 33.1 (MARINE DEVELOPMENT -- Space Utilization); 34.4 (SPACE DEVELOPMENT -- Communication); 41.5 (MATERIALS -- Electric Wires & Cables); 44.1 (COMMUNICATION -- Transmission Circuits & Antennae); 44.4 (COMMUNICATION -- Telephone); 44.6 (COMMUNICATION -- Television)

ABSTRACT

PROBLEM TO BE SOLVED: To have excellent high speed and operation stability and high adaptive performance and to enable an acoustic control, always maintaining large acoustic echo canceling amount by inserting a low - pass filter into a transmitting signal output terminal and interrupting

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the high frequency component of a transmitting signal when howling is detected.

SOLUTION: The device having the same constitution as an **acoustic echo removing device** 11 adopting a learning identifying method as an adaptive algorithm is composed of a howling detector 12, a band limit controller, a changeover switch 15 and a low-pass **filter** 16. When howling is detected by this howling detector 12, the band limit controller stops the successive update operation of a coefficient correction amount arithmetic circuit 7, operates the changeover switch 15, inserts the low-pass **filter** 16 and interrupts the high frequency component of a transmitting signal. Therefore, even if a communication line state is fallen into an unstable status, the generation of howling is suppressed without disconnecting the line and a speech state which is excellent in high speed and operation stability can be maintained.

12/5/3 (Item 3 from file: 347)

DIALOG(R)File 347:JAPIO
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02731494 **Image available**

ECHO ERASING DEVICE

PUB. NO.: 01-029094 [JP 1029094 A]
PUBLISHED: January 31, 1989 (19890131)
INVENTOR(s): OIKAWA HIROSHI
MAKINO SHOJI
MINAMI SHIGENOBU
SAEKI TAKASHI
APPLICANT(s): NIPPON TELEGR & TELEPH CORP <NTT> [000422] (A Japanese Company or Corporation), JP (Japan)
TOSHIBA CORP [000307] (A Japanese Company or Corporation), JP (Japan)
APPL. NO.: 62-183611 [JP 87183611]
FILED: July 24, 1987 (19870724)
INTL CLASS: [4] H04R-003/02; H04B-003/23; H04M-009/08
JAPIO CLASS: 42.5 (ELECTRONICS -- Equipment); 44.2 (COMMUNICATION -- Transmission Systems); 44.4 (COMMUNICATION -- Telephone)
JOURNAL: Section: E, Section No. 760, Vol. 13, No. 216, Pg. 163, May 19, 1989 (19890519)

ABSTRACT

PURPOSE: To effectively cancel an echo signal by assuming the reverberation time of a echo path from echo path characteristic assumed at each frequency band and variably setting a tap length in order to generate a pseudo echo signal in accordance with it.

CONSTITUTION: A **speaker** 1 and a microphone 2 are acoustically coupled. An input sound signal is separated by low-pass **filters** 11 and 21 and high-pass **filters** 12 and 22, the signals of respective frequency bands are sampled by frequency converting circuits 13, 23, 14 and 24, and the signals are supplied for the generation of the pseudo echo signal by an **echo canceler** circuit 3 for a **low - pass** and an **echo canceler** circuit 4 for a **high - pass**. The pseudo echo signal generated at the **echo canceler** circuit 3 for the **low - pass** and the **echo canceler** circuit 4 for the **high - pass** is supplied for the **cancelling** processing of the **echo** signal at subtracters 5 and 6.

12/5/4 (Item 4 from file: 347)

DIALOG(R)File 347:JAPIO
(c) 2003 JPO & JAPIO. All rts. reserv.

02313329 **Image available**

ECHO ELIMINATOR

June 27, 2003

PUB. NO.: 62-230229 [JP 62230229 A]
PUBLISHED: October 08, 1987 (19871008)
INVENTOR(s): MINAMI SHIGENOBU
APPLICANT(s): TOSHIBA CORP [000307] (A Japanese Company or Corporation), JP
(Japan)
APPL. NO.: 61-073411 [JP 8673411]
FILED: March 31, 1986 (19860331)
INTL CLASS: [4] H04B-003/20; H04M-009/08
JAPIO CLASS: 44.2 (COMMUNICATION -- Transmission Systems); 44.4
(COMMUNICATION -- Telephone)
JOURNAL: Section: E, Section No. 594, Vol. 12, No. 99, Pg. 140, March
31, 1988 (19880331)

ABSTRACT

PURPOSE: To eliminate noise produced around the boundary of each band by dividing the band into plural numbers, converting each band into a low frequency, eliminating an echo signal, restoring the frequency into the original band so as to eliminate the echo signal of all bands.

CONSTITUTION: When a transmission/reception discrimination circuit 9 discriminates the mode as the reception mode, the circuit 9 throws a switch 11 to the position B and throws a switch 13 to the position D. As a result, a reception signal inputted from a terminal 23 is outputted from a **speaker** 17 via a **band stop filter** 5, a band split **echo canceller** 3 and an amplifier 15. In this case, the band stop **filter** 5 eliminates the frequency component of the band near the boundary of the band to be split by the band split echo canceller 3, then the noise around it is eliminated by using a quadrature mirror **filter**. The noise is eliminated almost similarly in case of the transmission.

12/5/5 (Item 1 from file: 350)

DIALOG(R) File 350:Derwent WPIX
(c) 2003 Thomson Derwent. All rts. reserv.

014819847 **Image available**

WPI Acc No: 2002-640553/200269

XRPX Acc No: N02-506457

Echo canceller for full-duplex communication device e.g. modem, has sub-band echo canceller which suppresses band limited pseudo echo signal, in addition to adaptive type filter

Patent Assignee: RICOH KK (RICO)

Number of Countries: 001 Number of Patents: 001

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
JP 2002232329	A	20020816	JP 200121979	A	20010130	200269 B

Priority Applications (No Type Date): JP 200121979 A 20010130

Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
JP 2002232329	A	7		H04B-003/23	

Abstract (Basic): JP 2002232329 A

NOVELTY - An adaptive type **filter** suppresses a pseudo echo signal, using transmitted signal and received echo signal. A sub-band **echo canceller** (15) suppresses the **band limited** pseudo echo signal.

USE - In full-duplex communication device e.g. modem, **speaker** phone.

ADVANTAGE - Since band limited pseudo echo signals is suppressed, problem of frequency resolution is avoided.

DESCRIPTION OF DRAWING(S) - The figure shows the block diagram of the echo canceller. (Drawing includes non-English language text).

Sub-band echo canceller (15)

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pp; 7 DwgNo 1/5
Title Terms: ECHO; CANCEL; FULL; DUPLEX; COMMUNICATE; DEVICE; MODEM; SUB;
BAND; ECHO; CANCEL; SUPPRESS; BAND; LIMIT; PSEUDO; ECHO; SIGNAL; ADD;
ADAPT; TYPE; **FILTER**
Derwent Class: W02
International Patent Class (Main): H04B-003/23
File Segment: EPI

12/5/6 (Item 2 from file: 350)
DIALOG(R) File 350:Derwent WPIX
(c) 2003 Thomson Derwent. All rts. reserv.

010309586 **Image available**
WPI Acc No: 1995-210844/199528
XRPX Acc No: N95-165472

Echo canceller for teleconferencing system - has band division type low pass echo canceller that eliminates remaining echo in adaptive filter after pseudo echo derived from decimeter low pass data is subtracted from second

Patent Assignee: RICOH KK (RICO)
Number of Countries: 001 Number of Patents: 001

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
JP 7123028	A	19950512	JP 93265305	A	19931025	199528 B

Priority Applications (No Type Date): JP 93265305 A 19931025

Patent Details:

Patent No	Kind	Lan Pg	Main IPC	Filing Notes
JP 7123028	A	7	H04B-003/23	

Abstract (Basic): JP 7123028 A

The echo canceller has an adaptive **filter** which uses two decimeters (5.1,6.1) to generate low pass data. The voice signal received through the voice signal noise correcting microphone (12) is amplified, digitised and modulated with twice the sampling frequency before it is fed to the low frequency echo canceller (8) operating in the 7 KHz band. The echo canceller has a 3.4 KHz sampling frequency.

The digital voice data is transmitted through the codec (2) and the communication controller (2). The digital voice data received as a response is received by the same communication controller. The operation is reversed and the processed voice data is amplified to the **speaker**.

ADVANTAGE - Prevents echo generation. Eliminates howling. Increases device versatility since it can be used for both wideband and narrowband communications.

Dwg.1/2

Title Terms: ECHO; CANCEL; TELECONFERENCE; SYSTEM; BAND; DIVIDE; TYPE; LOW; PASS; ECHO; CANCEL; ELIMINATE; REMAINING; ECHO; ADAPT; **FILTER**; AFTER; PSEUDO; ECHO; DERIVATIVE; LOW; PASS; DATA; SUBTRACT; SECOND

Index Terms/Additional Words: ECHO; CANCEL; TELECONFEREN

Derwent Class: W01; W02

International Patent Class (Main): H04B-003/23

International Patent Class (Additional): H04M-009/08

File Segment: EPI

14/5/1 (Item 1 from file: 350)
DIALOG(R)File 350:Derwent WPIX
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013980443 **Image available**

WPI Acc No: 2001-464657/200150

Related WPI Acc No: 2001-343016

XRPX Acc No: N01-344652

Feedback canceling method for acoustic system, involves using least mean square algorithm for generating filter coefficients and providing low frequency input for LMS algorithm

Patent Assignee: OTICON AS (OTIC-N)

Inventor: EKELID M; NIELSEN J

Number of Countries: 024 Number of Patents: 003

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
WO 200106812	A1	20010125	WO 2000DK380	A	20000707	200150 B
AU 200058064	A	20010205	AU 200058064	A	20000707	200150
EP 1203510	A1	20020508	EP 2000943695	A	20000707	200238
			WO 2000DK380	A	20000707	

Priority Applications (No Type Date): DK 991043 A 19990719

Patent Details:

Patent No Kind Lan Pg Main IPC Filing Notes

WO 200106812 A1 E 23 H04R-025/00

Designated States (National): AU BR CA JP US

Designated States (Regional): AT BE CH CY DE DK ES FI FR GB GR IE IT LU MC NL PT SE

AU 200058064 A G10L-021/02 Based on patent WO 200106812

EP 1203510 A1 E H04R-025/00 Based on patent WO 200106812

Designated States (Regional): AT BE CH CY DE DK ES FI FR GB GR IE IT LI LU MC NL PT SE

Abstract (Basic): WO 200106812 A1

NOVELTY - The feedback canceling method involves using LMS (8) algorithm for generating filter coefficients (9) and using a high pass filter (20) to prevent low frequency signals from entering the LMS algorithm. The low frequency input for the LMS algorithm is provided by using the additional feedback cancellation filter (7) and a noise generator.

DETAILED DESCRIPTION - An INDEPENDENT CLAIM is also included for hearing aid.

USE - For acoustic system to cancel feedback.

ADVANTAGE - The method improves adaptation speed and eliminates side effects by fast suppression of feedback oscillations. The user comfort is improved by stabilizing the feedback cancellation and providing reliable coefficients for feedback canceling filter.

DESCRIPTION OF DRAWING(S) - The figure shows the schematic diagram of feedback canceling system.

Additional feedback cancellation filter (7)

LMS (8)

Filter coefficients (9)

High pass filter (20)

pp; 23 DwgNo 1/3

Title Terms: FEEDBACK; METHOD; ACOUSTIC; SYSTEM; MEAN; SQUARE; ALGORITHM; GENERATE; FILTER ; COEFFICIENT; LOW; FREQUENCY; INPUT; ALGORITHM

Derwent Class: P86; W04

International Patent Class (Main): G10L-021/02; H04R-025/00

File Segment: EPI; EngPI

14/5/2 (Item 2 from file: 350)

DIALOG(R)File 350:Derwent WPIX

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008689451

WPI Acc No: 1991-193471/199126

XRPX Acc No: N91-148132

Programmable hybrid hearing aid with signal processing - has open connection which constitute acoustic transmission channel with low - pass characteristic and resonant amplification

Patent Assignee: NHA AS (NHAN-N)

Inventor: KROKSTAD A; RAMSTAD T A; SVEAN J; RAMSTAD T

Number of Countries: 033 Number of Patents: 013

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
WO 9108654	A	19910613				199126 B
NO 8904806	A	19910531				199131
AU 9168805	A	19910626				199139
FI 9202408	A	19920526	WO 90NO178 FI 922408	A	19901129	199235
EP 502073	A1	19920909	WO 90NO178 EP 91900061	A	19901129	199237
JP 5504029	W	19930624	WO 90NO178 JP 91500704	A	19901129	199330
HU 63726	T	19930928	WO 90NO178 HU 921417	A	19901129	199344
US 5276739	A	19940104	WO 90NO178 US 92852242	A	19901129	199402
EP 502073	B1	19940914	WO 90NO178 EP 91900061	A	19901129	199435
DE 69012582	E	19941020	DE 612582 WO 90NO178 EP 91900061	A	19901129	199441
AU 654266	B	19941103	AU 9168805	A	19901129	199501
ES 2060345	T3	19941116	EP 91900061	A	19901129	199501
CA 2069737	C	19990914	CA 2069737 WO 90NO178	A	19901129	200004
				A	19901129	

Priority Applications (No Type Date): NO 894806 A 19891130

Cited Patents: 00 32690500; 00 33554200; 00 36403700; 00 4025900; 4187413

Patent Details:

Patent No Kind Lan Pg Main IPC Filing Notes

WO 9108654	A					
		Designated States (National): AT AU BB BG BR CA CH DE DK ES FI GB HU JP KP KR LK LU MC MG MW NO RO SD SE SU				
		Designated States (Regional): AT BE CH DE DK ES FR GB GR IT LU NL SE US				
CA 2069737	C E	H04R-025/00	Based on patent WO 9108654			
EP 502073	A1 E 10	H04R-025/00	Based on patent WO 9108654			
		Designated States (Regional): AT BE CH DE DK ES FR GB GR IT LI LU NL SE				
JP 5504029	W	H04R-025/00	Based on patent WO 9108654			
HU 63726	T	H04R-025/00	Based on patent WO 9108654			
US 5276739	A 19	H04R-025/00	Based on patent WO 9108654			
EP 502073	B1 E 35	H04R-025/00	Based on patent WO 9108654			
		Designated States (Regional): AT BE CH DE DK ES FR GB GR IT LI LU NL SE				
DE 69012582	E	H04R-025/00	Based on patent EP 502073 Based on patent WO 9108654			
AU 654266	B	H04R-025/00	Previous Publ. patent AU 9168805 Based on patent WO 9108654			
ES 2060345	T3	H04R-025/00	Based on patent EP 502073			
FI 9202408	A	H04R				

Abstract (Basic): WO 9108654 A

The programmable hearing aid comprises a main section (1) and two secondary sections (2a,2b) which are connected to the main section together with a battery.

The open connection of the aid constitutes an acoustic transmission channel with low - pass characteristic and resonant amplification . The hearing aid also includes an analog input

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section with a microphone amplifier (11) and a deconvolution **filter** (13). A digital signal processor with a compressor (33) and an equaliser (34) are also provided each of which contains RAM memories together with an analog output section with a reconstruction **filter** (14).

USE/ADVANTAGE - **Hearing aid** permits utilisation of hearing residue in bass range, and user can choose one of different response functions stored in **hearing aid** according to **acoustic environment**. (Dwg.No. 1a/6)

Title Terms: PROGRAM; HYBRID; HEARING; AID; SIGNAL; PROCESS; OPEN; CONNECT; CONSTITUTE; ACOUSTIC; TRANSMISSION; CHANNEL; LOW; PASS; CHARACTERISTIC; RESONANCE; AMPLIFY

Derwent Class: W04

International Patent Class (Main): H04R-007/185; **H04R-025/00**

International Patent Class (Additional): H04R-025/02

File Segment: EPI

14/5/3 (Item 3 from file: 350)

DIALOG(R) File 350:Derwent WPIX

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008230324 **Image available**

WPI Acc No: 1990-117325/199016

XRPX Acc No: N90-090953

Integrated compression amplifier with programmable threshold voltage - has feedback circuit containing rectifier and low - pass filter connected to control input of 2-quadrant amplifier

Patent Assignee: SIEMENS AG (SIEI)

Inventor: MAUTHE M

Number of Countries: 012 Number of Patents: 007

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
EP 363714	A	19900418	EP 89117648	A	19890925	199016 B
CA 2000434	A	19900413				199019
JP 2149111	A	19900607	JP 89266057	A	19891011	199029
US 4987383	A	19910122	US 89412166	A	19890925	199106
CA 2000434	C	19931207	CA 2000434	A	19891011	199404
EP 363714	B1	19950405	EP 89117648	A	19890925	199518
DE 58909157	G	19950511	DE 509157	A	19890925	199524
			EP 89117648	A	19890925	

Priority Applications (No Type Date): DE 3834928 A 19881013

Cited Patents: 3.Jnl.Ref; A3...9041; NoSR.Pub; US 3919654; US 4539440

Patent Details:

Patent No Kind Lan Pg Main IPC Filing Notes

EP 363714 A Designated States (Regional): AT CH DE FR GB IT LI NL

EP 363714 B1 G 32 H03G-009/02

Designated States (Regional): AT CH DE FR GB IT LI NL

DE 58909157 G H03G-009/02 Based on patent EP 363714

CA 2000434 C H03F-001/38

Abstract (Basic): EP 363714 A

The amplifier has a 2-quadrant multiplier (2M) and a **feedback loop** containing a rectifier stage (GR) and a **low - pass filter** (TP) providing an output signal which is supplied to the control input of the 2-quadrant multiplier (2M). The output of the latter is coupled via a separation amplifier (TV) to a controlled amplification stage (SC).

The rectifier stage (GR) is coupled to a bias voltage generator (GV), with the output of the rectifier stage (GR) coupled to the **low - pass filter** (TP) via a summation point (S) receiving an output current (IO) from a reference current source (RI).

USE - For multi-channel automatic gain controller in **hearing aid**

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. (22pp Dwg.No.4/9)
Title Terms: INTEGRATE; COMPRESS; AMPLIFY; PROGRAM; THRESHOLD; VOLTAGE;
FEEDBACK; CIRCUIT; CONTAIN; RECTIFY; LOW; PASS; FILTER ; CONNECT;
CONTROL; INPUT; QUADRANT; AMPLIFY

Derwent Class: U24

International Patent Class (Main): H03F-001/38; H03G-009/02

International Patent Class (Additional): H03G-003/12; H03G-007/08;
H03G-009/12; **H04R-025/00**

File Segment: EPI

14/5/4 (Item 4 from file: 350)

DIALOG(R) File 350:Derwent WPIX

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008213435 **Image available**

WPI Acc No: 1990-100436/199014

XRPX Acc No: N90-077626

Hearing aid system - has feedback loop to control variable
filter to reduce effect of noise

Patent Assignee: BELTONE ELECTRONICS CORP (BELT-N)

Inventor: ANDERSON J R

Number of Countries: 004 Number of Patents: 005

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
DE 3927765	A	19900329	DE 3927765	A	19890823	199014 B
FR 2635680	A	19900302				199016
JP 2113698	A	19900425	JP 89221887	A	19890830	199023
US 5170434	A	19921208	US 88238207	A	19880830	199252
			US 89459309	A	19891229	
			US 91722926	A	19910628	
DE 3927765	C2	19930527	DE 3927765	A	19890823	199321

Priority Applications (No Type Date): US 88238207 A 19880830; US 89459309 A
19891229; US 91722926 A 19910628

Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
DE 3927765	A		8		
US 5170434	A		6	H04R-025/00	Cont of application US 88238207 Cont of application US 89459309
DE 3927765	C2		8	H04R-025/00	

Abstract (Basic): DE 3927765 A

A hearing aid has a microphone (12), a variable filter (14),
amplifier (16) and a sensor unit (18). The microphone receives speech
input that is passed through the filter which is a high pass
device with the cut off frequency determined by the control input (20).
The sensor circuit has a threshold level control (25), band pass
filter (26), level detector (30) and a smoothing circuit (32).

The band pass filter has a centre frequency of 250 Hz and
generates an output that is interpreted by the detector to identify
noise. This results in the variable filter (14) being adjusted to
effectively reduce the noise effect.

ADVANTAGE - Modifies signal to reduce received noise effect

Title Terms: HEARING; AID; SYSTEM; FEEDBACK; LOOP; CONTROL; VARIABLE;
FILTER ; REDUCE; EFFECT; NOISE

Derwent Class: P32; U25; W04

International Patent Class (Main): **H04R-025/00**

International Patent Class (Additional): A61F-011/04

File Segment: EPI; EngPI

14/5/5 (Item 5 from file: 350)

DIALOG(R) File 350:Derwent WPIX

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003763649

WPI Acc No: 1983-759861/198337

XRPX Acc No: N83-160439

Amplifier with automatic gain control for hearing aid - has variable resistor and two transistors in feedback loop

Patent Assignee: AUDIBEL (AUDI-N); PHILIPS GLOEILAMPENFAB NV (PHIG)

Inventor: RIDEL P

Number of Countries: 007 Number of Patents: 009

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week	
DE 3306441	A	19830908				198337	B
FR 2522451	A	19830902				198340	
GB 2117200	A	19831005				198340	
JP 58162115	A	19830926				198344	
DK 8300939	A	19831107				198351	
US 4509022	A	19850402	US 83470745	A	19830228	198516	
GB 2117200	B	19851211				198550	
IT 1167626	B	19870513				198941	
DE 3306441	C	19910822				199134	

Priority Applications (No Type Date): FR 823347 A 19820301

Patent Details:

Patent No	Kind	Lan Pg	Main IPC	Filing Notes
DE 3306441	A	8		

Abstract (Basic): DE 3306441 A

The amplifier has an input transducer, an amplifying unit (30) an output transducer and automatic gain control. A variable resistor (40) is used to obtain an ac voltage having the same phase as the signal from the input transducer. A rectifier (50) and an RC filter are provided.

The filter's output is connected to the base of a first transistor (70) whose collector controls the base of a second transistor (80). The second transistor is connected to the input of the amplifier unit such that it increases the short circuiting experienced by this unit's input signal as the signal from the variable resistor becomes larger. The advantage lies in the amplifier's being suitable for the small supply voltages (c. 1.3V) found in hearing aids .

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Title Terms: AMPLIFY; AUTOMATIC; GAIN; CONTROL; HEARING; AID; VARIABLE; RESISTOR; TWO; TRANSISTOR; FEEDBACK; LOOP

Derwent Class: U24; W04

International Patent Class (Additional): H03G-003/20; H03G-007/06; H04R-025/00

File Segment: EPI

14/5/6 (Item 6 from file: 350)

DIALOG(R) File 350:Derwent WPIX

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003613523

WPI Acc No: 1983-G1722K/198319

XRPX Acc No: N83-080160

hearing aid amplifier circuit - has filter with wide-band and high - pass filter paths

Patent Assignee: SIEMENS AG (SIEI)

Inventor: SCHLOSSER H

Number of Countries: 001 Number of Patents: 002

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week	
DE 3141420	A	19830505	DE 3141420	A	19811019	198319	B
DE 3141420	C	19890202				198905	

June 27, 2003

Priority Applications (No Type Date): DE 3141420 A 19811019

Patent Details:

Patent No	Kind	Lan Pg	Main IPC	Filing Notes
DE 3141420	A	16		

Abstract (Basic): DE 3141420 A

The amplifier circuit (6,7) is inserted between a microphone and a hearing capsule, for amplifying the received sound waves. It incorporates a volume control and a frequency **filter** (7), the latter exhibiting two signal paths (7a,7b) with wideband and **high pass filter** characteristics respectively. The signals fed along the two signal paths (7a,7b) exhibiting a relative phase shift of about 180 degrees.

The wideband signal path (7a) has a variable resistor (24) and contains only passive components so that it can transmit signals in either direction for simultaneously acting as a **feedback loop**. The other signal path (7b) has a positive amplification of between 5 and 10dB. The amplifier circuit allows optimum sound balancing.

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Title Terms: HEARING; AID; AMPLIFY; CIRCUIT; **FILTER**; WIDE; BAND; HIGH; PASS; **FILTER**; PATH

Derwent Class: U25; W04

International Patent Class (Additional): H04R-003/04; **H04R-025/00**

File Segment: EPI

June 27, 2003

15/5/1 (Item 1 from file: 347)

DIALOG(R) File 347:JAPIO

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05449993 **Image available**

LOUD SPEAKER INFORMATION COMMUNICATION SYSTEM

PUB. NO.: 09-064793 [JP 9064793 A]

PUBLISHED: March 07, 1997 (19970307)

INVENTOR(s): IWASAKI TAKASHI

KUSANO YOSHIMASA

APPLICANT(s): KYOCERA CORP [358923] (A Japanese Company or Corporation), JP (Japan)

APPL. NO.: 07-211752 [JP 95211752]

FILED: August 21, 1995 (19950821)

INTL CLASS: [6] H04B-003/23; H03H-021/00; H04M-001/60

JAPIO CLASS: 44.2 (COMMUNICATION -- Transmission Systems); 33.1 (MARINE DEVELOPMENT -- Space Utilization); 34.4 (SPACE DEVELOPMENT -- Communication); 41.5 (MATERIALS -- Electric Wires & Cables); 44.1 (COMMUNICATION -- Transmission Circuits & Antennae); 44.4 (COMMUNICATION -- Telephone); 44.6 (COMMUNICATION -- Television)

ABSTRACT

PROBLEM TO BE SOLVED: To have excellent high speed and operation stability and high adaptive performance and to enable an acoustic control, always maintaining large acoustic **echo canceling** amount by inserting circuit loss into the line of a high frequency band and interrupting the high frequency component of a transmitting signal when howling is detected.

SOLUTION: This system is composed of an **acoustic echo removing device** 11 adopting a learning identifying method as adaptive algorithm, an analysis **filter** 12, a synthetic **filter** 13, a down-sampling circuit 14, an up-sampling circuit 15, a recoupling addition circuit 16, a howling detector 17, a **band limit** controller 18 and a circuit loss control circuit 19. When howling is detected by the howling detector 17, the successive update operation of a coefficient correction amount arithmetic circuit 7 on a high frequency band side is stopped, circuit loss is inserted into a pertinent high frequency band line and the high frequency component of a transmitting signal is interrupted. Thus, the probability of the generation of howling can be reduced.

15/5/2 (Item 2 from file: 347)

DIALOG(R) File 347:JAPIO

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04604263 **Image available**

DIGITAL AUDIO TRANSMITTING DEVICE AND RECEIVING DEVICE.

PUB. NO.: 06-276163 [JP 6276163 A]

PUBLISHED: September 30, 1994 (19940930)

INVENTOR(s): KATSUMATA TORU

MATSUI JO

APPLICANT(s): SONY CORP [000218] (A Japanese Company or Corporation), JP (Japan)

APPL. NO.: 05-088116 [JP 9388116]

FILED: March 23, 1993 (19930323)

INTL CLASS: [5] H04B-014/04; H04N-007/13

JAPIO CLASS: 44.2 (COMMUNICATION -- Transmission Systems); 44.6 (COMMUNICATION -- Television)

JOURNAL: Section: E, Section No. 1651, Vol. 18, No. 687, Pg. 86; December 26, 1994 (19941226)

ABSTRACT

June 27, 2003

PURPOSE: To obtain digital audio transmitting and receiving devices whereby a circuit scale is made to be small by unifying the sampling frequency of an encoding system with different sampling frequencies.

CONSTITUTION: At the time of transmission, only the prescribed band of a signal supplied to a digital **low pass filter** 24 is supplied to a 1/2 thinning circuit 25 in a band converting part 2. Though the actual sampling frequency is reduced by the circuit 25, the **filter** 24 is provided before the circuit for satisfying a sampling theorem even at that time so that the band is limited. In the meantime, at the time of reception, a reception signal inputted to a signal separating part 31 with a digital line interface 30 is separated into a video signal and a sign code. The digital audio signal from a telephone quality voice decoder 32 is supplied to a sampling frequency converting part 10 and converted into the operation frequency of **echo canceller** 23. Thus, the audio signal is made to be the sampling frequency being the same as the audio signal from a high quality voice decoding part 36.

15/5/3 (Item 3 from file: 347)

DIALOG(R)File 347:JAPIO

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04064261 **Image available**

LOUD- **SPEAKER** SIMULTANEOUS SPEECH SYSTEM USING FREQUENCY DIVISION

PUB. NO.: 05-055961 [JP 5055961 A]

PUBLISHED: March 05, 1993 (19930305)

INVENTOR(s): ISHIKAWA KATSUKI

APPLICANT(s): AIPHONE CO LTD [324020] (A Japanese Company or Corporation),
JP (Japan)

APPL. NO.: 03-215459 [JP 91215459]

FILED: August 27, 1991 (19910827)

INTL CLASS: [5] H04B-003/23; H04M-001/60

JAPIO CLASS: 44.2 (COMMUNICATION -- Transmission Systems); 44.4
(COMMUNICATION -- Telephone)

JAPIO KEYWORD: R131 (INFORMATION PROCESSING -- Microcomputers &
Microprocessors)

JOURNAL: Section: E, Section No. 1395, Vol. 17, No. 359, Pg. 63, July
07, 1993 (19930707)

ABSTRACT

PURPOSE: To make the operation stable in double talking over the entire band of a voice signal by deciding the speech band respectively for the **echo canceller** system for reducing incoming and outgoing voice signals and for the system dividing the frequency depending on a cause of call occurrence for a high frequency so as to improve an **echo cancel** function.

CONSTITUTION: A low frequency component of a voice band at a caller side separated by a **low pass filter** 4 of a loudspeaking simultaneous speech equipment TE(sub 1) is converted into a digital signal and it is sent to a CPU 19. An **echo cancel** arithmetic operation program built in the CPU 19 references an outgoing voice signal from the **low pass filter** 11 to calculate a simulating echo. An incoming voice signal FS from which the simulating echo is subtracted is sent to a speech use D/A converter 6. On the other hand, a high frequency component in the incoming voice signal sent from an incoming caller side low frequency elimination **filter** 2 is eliminated in a way of an interdigital **filter** corresponding to the frequency of the outgoing voice signal FS of the caller **band pass filter** 9 so as to prevent howling of the high frequency component.

15/5/4 (Item 4 from file: 347)

DIALOG(R)File 347:JAPIO

June 27, 2003

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02252013 **Image available**
EXHAUST SOUND REDUCING DEVICE FOR ENGINE

PUB. NO.: 62-168913 [JP 62168913 A]
PUBLISHED: July 25, 1987 (19870725)
INVENTOR(s): INOUE HIROSHI
APPLICANT(s): MAZDA MOTOR CORP [000313] (A Japanese Company or Corporation), JP (Japan)
APPL. NO.: 61-010507 [JP 8610507]
FILED: January 20, 1986 (19860120)
INTL CLASS: [4] F01N-001/00
JAPIO CLASS: 21.2 (ENGINES & TURBINES, PRIME MOVERS -- Internal Combustion); 32.9 (POLLUTION CONTROL -- Other)
JOURNAL: Section: M, Section No. 657, Vol. 12, No. 4, Pg. 84, January 08, 1988 (19880108)

ABSTRACT

PURPOSE: To prevent the acoustic **feedback** effect for reducing efficiently exhaust sounds, by entering a signal, which has a burning noise frequency corresponding to engine speed and a phase opposite to exhaust pulsation, into a pulsation generator.

CONSTITUTION: On an exhaust passage 1, a pressure sensor 2 is provided, and the signal from the sensor 2 is input in a sequential type probability control system 8 via a **band - pass filter** 3 and an A/D converter 4. Further, the signal from an engine speed sensor 5 is entered in a frequency converter 6, and in the converter 6 the frequency of engine speed pulse is converted to a basic frequency of burning sound. And, in the control system 8, a signal having a burning sound frequency and a phase opposite to exhaust pulsation is formed, and sent to a **speaker** 11. Thus, the acoustic **feedback** effect is **prevented**, and exhaust sounds can be reduced.

15/5/5 (Item 5 from file: 347)

DIALOG(R)File 347:JAPIO
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01907524 **Image available**
ECHO CONTROL SYSTEM

PUB. NO.: 61-121624 [JP 61121624 A]
PUBLISHED: June 09, 1986 (19860609)
INVENTOR(s): UMIGAMI SHIGEYUKI
MURANO KAZUO
KOSHIKAWA MASAMI
APPLICANT(s): FUJITSU LTD [000522] (A Japanese Company or Corporation), JP (Japan)
APPL. NO.: 59-243808 [JP 84243808]
FILED: November 19, 1984 (19841119)
INTL CLASS: [4] H04B-003/20
JAPIO CLASS: 44.2 (COMMUNICATION -- Transmission Systems)
JOURNAL: Section: E, Section No. 447, Vol. 10, No. 309, Pg. 65, October 21, 1986 (19861021)

ABSTRACT

PURPOSE: To realize an **echo canceller** with simplified circuit constitution by splitting a transmission system into a low-frequency signal component and a high-frequency signal component and using the **echo canceller** for the low frequency signal component to erase the echo component and using a comb-line **filter** with respect to the high frequency signal component to extract the frequency.

CONSTITUTION: After an incoming signal is converted into a digital signal,

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the result is separated into high and low frequency signals by a **band pass** separation **filter** 12 of an incoming system. The separated high frequency component is subjected to spectral interleaving by a comb-line **filter** 13 extracting the frequency component of a prescribed interval. The interleaving signal is added to the separated low frequency component at an adder **filter** 14 to drive a **speaker** 3. The sound signal of a microphone 1 is separated into the high and low frequency signals by a **band pass** **filter** 8 the same as that of the incoming system. The echo component is separated into the high and low frequencies, and the echo is blocked in terms of frequencies with respect to the high frequency component by the comb-line **filter** 10. The echo is eliminated with respect to the low frequency component by using a well-known **echo canceller** 9.

15/5/6 (Item 6 from file: 347)
DIALOG(R)File 347:JAPIO
(c) 2003 JPO & JAPIO. All rts. reserv.

00413022
DRIVING CIRCUIT SYSTEM OF MULTIWAY **SPEAKERS**

PUB. NO.: 54-065022 [JP 54065022 A]
PUBLISHED: May 25, 1979 (19790525)
INVENTOR(s): TAKAHASHI NOBUAKI
 FUNASAKA EIICHI
 SHINOZAKI MASANOBU
 KAIZU YASUO
APPLICANT(s): VICTOR CO OF JAPAN LTD [000432] (A Japanese Company or Corporation), JP (Japan)
APPL. NO.: 52-131010 [JP 77131010]
FILED: November 01, 1977 (19771101)
INTL CLASS: [2] H04R-003/12; H03F-001/34
JAPIO CLASS: 42.5 (ELECTRONICS -- Equipment); 42.4 (ELECTRONICS -- Basic Circuits)
JOURNAL: Section: E, Section No. 125, Vol. 03, No. 86, Pg. 71, July 24, 1979 (19790724)

ABSTRACT

PURPOSE: To improve the damping characteristics of multiway **speakers** by adding and synthesizing plural signals having undergone frequency division and negative- feedback-operating this synthesized signal.

CONSTITUTION: The output signal of an amplifier 2 is applied to a tweeter 3 through a **high - pass filter**, to a squawker 4 through a **band - pass filter** and to a woofer 5 through a **low - pass filter**. The signals being applied to the tweeter 3, squawker 4 and woofer 5 after having been frequency-divided are respectively synthesized through resistances R_(sub 1), R_(sub 2), R_(sub 3) and the synthesized signal is voltage-divided in resistances R_(sub 4), R_(sub 5), after which it is applied to the inversion input terminal of the amplifier 2, whereby negative feedback is applied thereto. Then, even those up to the signals driving the **speakers** 3, 4, 5 are included in the negative **feedback loop** and therefore, the degradation in the damping factors owing to the presence of equivalent series resistances in capacitors C_(sub 1), C_(sub 2), coils L_(sub 1), L_(sub 2) may be eliminated.

15/5/7 (Item 1 from file: 350)
DIALOG(R)File 350:Derwent WPIX
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014803722 **Image available**
WPI Acc No: 2002-624428/200267
Noise and echo canceling apparatus for use in hands free kit includes a high pass filter and a low pass filter sequentially

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filter out non-audible frequency components of the user's voice signal from the microphone

Patent Assignee: AEROTELECOM CO LTD (AERO-N); AERO TELECOM CO LTD (AERO-N)

Inventor: SEO G I; SUH G I

Number of Countries: 001 Number of Patents: 002

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
KR 2002012028	A	20020215	KR 200045447	A	20000805	200267 B
KR 335404	B	20020506	KR 200045447	A	20000805	200271

Priority Applications (No Type Date): KR 200045447 A 20000805

Patent Details:

Patent No Kind Lan Pg Main IPC Filing Notes

KR 2002012028 A 1 H04R-003/02

KR 335404 B H04R-003/02 Previous Publ. patent KR 2002012028

Abstract (Basic): KR 2002012028 A

NOVELTY - The ground(7) is provided at a microphone(5) powered from the power supply(3) to ground a user's voice signal before transferring it to a noise canceler (6). The ground (9) connected to a speaker (8) to ground the other party's voice signal from the noise canceler(6). A high pass filter and a low pass filter sequentially filter out non-audible frequency components of the user's voice signal from the microphone (5). An analog switch turns on/off the output of the user's voice signal depending on the other party's voice signal.

DETAILED DESCRIPTION - In noise and echo canceling apparatus for a hands free, a power supply ground(4), a microphone ground(7) and a speaker ground(9) are separated from each other. The ground (7) is connected to a power supply (3) receiving power through a plug coupled to a cigarette lighter of a car to ground charge noise without effecting on other circuits.

USE - For hands-free apparatus of a GSM handheld phone.

ADVANTAGE - The noise and echo canceling is achieved by reducing effect of charge noise and grounds, amplifying audible frequencies only, and, if no signals at a speaker, outputting user's voice.

pp; 1 DwgNo 1/10

Title Terms: NOISE; ECHO; APPARATUS; HAND; FREE; KIT; HIGH; PASS; FILTER ; LOW; PASS; FILTER ; SEQUENCE; FILTER ; NON; AUDIBLE; FREQUENCY; COMPONENT; USER; VOICE; SIGNAL; MICROPHONE

Derwent Class: P86; U25; W01

International Patent Class (Main): H04R-003/02

International Patent Class (Additional): G10K-011/00

File Segment: EPI; EngPI

15/5/8 (Item 2 from file: 350)

DIALOG(R) File 350:Derwent WPIX

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012386052 **Image available**

WPI Acc No: 1999-192159/199917

XRPX Acc No: N99-140734

Detection of direction of speech activity e.g. for hands-free speaker telephone

Patent Assignee: NOKIA MOBILE PHONES LTD (OYNO)

Inventor: HAEKKINEN J; VALVE P; IIPPONEN P

Number of Countries: 026 Number of Patents: 003

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
EP 901267	A2	19990310	EP 98660083	A	19980826	199917 B
FI 9703596	A	19990305	FI 973596	A	19970904	199924
JP 11168791	A	19990622	JP 98249875	A	19980903	199935

Priority Applications (No Type Date): FI 973596 A 19970904

June 27, 2003

Patent Details:

Patent No	Kind	Lan Pg	Main IPC	Filing Notes
EP 901267	A2	E	14 H04M-009/08	
Designated States (Regional): AL AT BE CH CY DE DK ES FI FR GB GR IE IT LI LT LU LV MC MK NL PT RO SE SI				
JP 11168791	A	42	H04R-003/00	
FI 9703596	A		H04B-000/00	

Abstract (Basic): EP 901267 A

NOVELTY - Microphones, preferably four or more, in a microphone vector (2) receive a voice signal and each produces a single signal. This is passed through **band - pass filters** (14) to direction angle estimating to units (15, 17), that store the assumed direction of arrival of the voice. This is then compared to the assumed direction in a detection unit (18), to identify a match of the assumed and estimated directions of arrival of a voice signal.

DETAILED DESCRIPTION - An INDEPENDENT CLAIM is included for a detection device of voice sources.

USE - Detecting source of voice using receiving microphone e.g. in full duplex **speaker** phones that are prone to echo.

ADVANTAGE - Achieves **echo - canceling** and suppresses information of about message-end speech activity

DESCRIPTION OF DRAWING(S) - The drawing is a block diagram of detector according to present invention.

Microphone vector 2

Band - pass filter 14

Estimation and recording units 15, 17

Direction detection unit 18

Dwg.2/9

Title Terms: DETECT; DIRECTION; SPEECH; ACTIVE; HAND; FREE; **SPEAKER** ;
TELEPHONE

Derwent Class: P86; W01; W04

International Patent Class (Main): H04B-000/00; H04M-009/08; H04R-003/00

International Patent Class (Additional): G01H-003/00; G10L-009/00;
H04M-001/60

File Segment: EPI; EngPI

15/5/9 (Item 3 from file: 350)

DIALOG(R) File 350:Derwent WPIX

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011468484 **Image available**

WPI Acc No: 1997-446391/199741

XRPX Acc No: N97-372012

Adaptive type noise removal appts for vehicle telephone - has adaptive filters , which vary characteristics of each signal path of transmission signal to which presumed echo is added, adaptively based on adder's output

Patent Assignee: NEC CORP (NIDE)

Number of Countries: 001 Number of Patents: 001

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
JP 9205388	A	19970805	JP 9611574	A	19960126	199741 B

Priority Applications (No Type Date): JP 9611574 A 19960126

Patent Details:

Patent No	Kind	Lan Pg	Main IPC	Filing Notes
JP 9205388	A	6	H04B-003/23	

Abstract (Basic): JP 9205388 A

The appts consists of an acoustic **echo canceller** (12) between a radio (6) and a **speaker** (11). The output transmission signals from a microphone (1) is passed to the radio. The acoustic **echo canceller** forms a presumed **echo** based on the answering signal. Presumed echo is

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added to the transmission signal through an adder (4) and the acoustic echo component is removed from the transmission signal.

A first and second adaptive **high pass filters** (14,15) are provided at front stage of adder and adaptive **echo canceller** respectively. The variation of the characteristics of each signal path of the transmission signal is adaptively done by the adaptive **high pass filter**, based on the output of the adder.

ADVANTAGE - Improves S/N of transmission signal. Reduces acoustic echoes. Improves interactive quality and transmission articulation.

Dwg.1/5

Title Terms: ADAPT; TYPE; NOISE; REMOVE; APPARATUS; VEHICLE; TELEPHONE; ADAPT; **FILTER**; VARY; CHARACTERISTIC; SIGNAL; PATH; TRANSMISSION; SIGNAL ; ECHO; ADD; ADAPT; BASED; ADDER; OUTPUT

Derwent Class: U22; W01

International Patent Class (Main): H04B-003/23

International Patent Class (Additional): H03H-017/00; H03H-021/00;
H04M-001/60

File Segment: EPI

15/5/10 (Item 4 from file: 350)

DIALOG(R) File 350:Derwent WPIX

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010250988 **Image available**

WPI Acc No: 1995-152243/199520

XRPX Acc No: N95-119680

Noise reduction device for air conditioner - incorporates pair of speakers and microphone in ventilation duct to impart symmetry to acoustic characteristic of ventilation duct about vertical axis

Patent Assignee: MATSUSHITA DENKI SANGYO KK (MATU)

Number of Countries: 001 Number of Patents: 001

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
JP 7077994	A	19950320	JP 93222702	A	19930908	199520 B

Priority Applications (No Type Date): JP 93222702 A 19930908

Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
JP 7077994	A	13	G10K-011/178		

Abstract (Basic): JP 7077994 A

The noise reduction device has several **FIR filters**. Two microphones (1a,1b) and two **speakers** (5a,5b) are located in the ventilation duct (9) attached to the blower (8). The first microphone (1a) detects the acoustic noise emitted by the air blower. The LMS calculator (4) executes the signal processing of detected acoustic noise. The adaptive **filter** (2) executes the adaptive control of the calculator output.

The two **speakers** (5a,5b) generates acoustic output from the adaptive **filter** electrical output. The second digital **filter** (3b) processes the signal outputted by the adaptive **filter** and feeds the result to the LMS calculator (4). The second acoustic noise detector i.e. microphone is installed in the ventilation duct at far end near the **speaker**. The acoustic noise detected by the second microphone is fed to the LMS calculator (4). The **speakers** (5a,5b) are arranged so that the cones face each other, ensuring axial alignment with direction of airflow.

ADVANTAGE - Improves noise reduction effect and **low pass** reproduction capability with low cost **speakers**. Provides uniform and stable attenuation effect. Prevents feedback loop constituted by output signal of second digital **filter** and adaptive **filter** reproduced by calculator from **speaker** is detected by first microphone. Provides stable noise control. Delivers good acoustic damping. Regulates flow of air between first and second detection

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points.

Dwg.1/9

Title Terms: NOISE; REDUCE; DEVICE; AIR; CONDITION; INCORPORATE; PAIR;
SPEAKER; MICROPHONE; VENTILATION; DUCT; IMPART; SYMMETRICAL; ACOUSTIC;
CHARACTERISTIC; VENTILATION; DUCT; VERTICAL; AXIS

Derwent Class: P86; Q74; U22; W04; X27

International Patent Class (Main): G10K-011/178

International Patent Class (Additional): F24F-013/02; H03H-017/02;
H03H-021/00

File Segment: EPI; EngPI

15/5/11 (Item 5 from file: 350)

DIALOG(R) File 350:Derwent WPIX

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009041337 **Image available**

WPI Acc No: 1992-168696/199221

XRPX Acc No: N92-127146

RF signal transceiver for remote access to vehicles - uses RF oscillator stage with feedback loop incorporating saw delay line and control switching between transmitting and receiving modes

Patent Assignee: DELCO ELECTRONICS CORP (DELC-N)

Inventor: ANDERSON F J

Number of Countries: 006 Number of Patents: 009

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week	
EP 486087	A2	19920520	EP 91202848	A	19911101	199221	B
AU 9187030	A	19920611	AU 9187030	A	19911106	199232	
US 5146613	A	19920908	US 90614488	A	19901116	199239	
JP 4269014	A	19920925	JP 91300357	A	19911115	199245	
EP 486087	A3	19921125	EP 91202848	A	19911101	199343	
NZ 240600	A	19940427	NZ 240600	A	19911114	199420	
EP 486087	B1	19950906	EP 91202848	A	19911101	199540	
DE 69112774	E	19951012	DE 612774	A	19911101	199546	
			EP 91202848	A	19911101		
KR 9507493	B1	19950711	KR 9120448	A	19911116	199715	

Priority Applications (No Type Date): US 90614488 A 19901116

Cited Patents: No-SR.Pub; 1.Jnl.Ref; FR 2165740; US 4786903

Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
EP 486087	A2	E	6	H04B-001/40	
US 5146613	A		7	H04B-001/44	
JP 4269014	A		5	H04B-001/44	
EP 486087	B1	E	7	H04B-001/40	
DE 69112774	E			H04B-001/40	Based on patent EP 486087
AU 9187030	A			H04B-001/40	
EP 486087	A3			H04B-001/40	
NZ 240600	A			H04B-001/40	
KR 9507493	B1			H04B-001/44	

Abstract (Basic): EP 486087 A

The transceiver has an RF oscillator (20) which includes a feedback circuit (22,24) comprising a surface **acoustic** wave **device** (24) coupling an input and an output of the RF oscillator, and adapted to produce RF oscillations. A controller (14) switches the appts. between transmitting and receiving modes. A transmitter (38,40,12) is coupled to the output of the RF oscillator during the transmitting mode to transmit the RF oscillations produced.

An input (12,16,18) is adapted during the receiving node to couple a modulated RF signal to the input of the RF oscillator. A **low . pass filter** (34) is coupled to the output of the RF oscillator, and adapted to **filter** the modulated RF signal.

ADVANTAGE - Small size, low power consumption, and good high temp.

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stability.

Dwg.1-3/6

Title Terms: RF; SIGNAL; TRANSCEIVER; REMOTE; ACCESS; VEHICLE; RF;
OSCILLATOR; STAGE; FEEDBACK; LOOP; INCORPORATE; SAW; DELAY; LINE; CONTROL
; SWITCH; TRANSMIT; RECEIVE; MODE

Derwent Class: W05; X22

International Patent Class (Main): H04B-001/40; H04B-001/44

International Patent Class (Additional): H04B-001/30

File Segment: EPI

15/5/12 (Item 6 from file: 350)

DIALOG(R) File 350:Derwent WPIX

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008423815

WPI Acc No: 1990-310816/199041

XRPX Acc No: N90-238370

Extended bandwidth virtual earth noise controller - has filter between
speaker and microphone moved nearer noise source, so transport lead
component reduces loop gain requirements

Patent Assignee: ANONYMOUS (ANON)

Number of Countries: 001 Number of Patents: 001

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
RD 317102	A	19900910			199041	B

Priority Applications (No Type Date): RD 90317102 A 19900820

Abstract (Basic): RD 317102 A

The virtual earth active noise controller consists of a microphone for measuring acoustic noise generated by a source, an inverting amplifier having the microphone connected to its input, and a speaker connected to the output of the inverting amplifier. The noise signal measured by the input microphone is inverted by the amplifier, and corresponding acoustic waves are produced by the speaker to cancel the noise waves. The input microphone is moved a slight distance away from the speaker and toward the source of noise. A filtering network is included between the input microphone and speaker.

The measured signal produced by the input microphone then consists of a transport lead component due to the measured noise, and a loop feedback component due to the waves produced by the speaker. The transport lead component reduces the loop gain requirements for optimum noise reduction and stability. The total system noise attenuation is significantly greater than the loop gain and is attributable to the transport lead component. The filtering network is designed to provide the proper amplification and phase compensation, in accord with the transfer functions of the other system components, such as the microphone and speaker, to increase the loop gain at the bandpass limits.

ADVANTAGE - Operable noise canceling bandwidth of noise controller is significantly increased.

Dwg.0/0

Title Terms: EXTEND; BANDWIDTH; VIRTUAL; EARTH; NOISE; CONTROL; FILTER ;
SPEAKER ; MICROPHONE; MOVE; NEARBY; NOISE; SOURCE; SO; TRANSPORT; LEAD;
COMPONENT; REDUCE; LOOP; GAIN; REQUIRE

Derwent Class: P86; W04

International Patent Class (Additional): G10K-000/00

File Segment: EPI; EngPI

15/5/13 (Item 7 from file: 350)

DIALOG(R) File 350:Derwent WPIX

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004181455

WPI Acc No: 1985-008335/198502

XRPX Acc No: N85-005840

Sound attenuation system using active control techniques - has controller incorporating signal processor feeding signal from sound detector to generator using negative feedback

Patent Assignee: NAT RES DEV CORP (NATR)

Inventor: SWINBANKS M A

Number of Countries: 003 Number of Patents: 004

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
GB 2142091	A	19850109	GB 8415387	A	19840615	198502 B
JP 60020700	A	19850201	JP 84128374	A	19840621	198511
US 4589133	A	19860513	US 84620751	A	19840614	198622
GB 2142091	B	19870325				198712

Priority Applications (No Type Date): GB 8317086 A 19830623; GB 8415387 A 19840615

Patent Details:

Patent No	Kind	Lan Pg	Main IPC	Filing Notes
GB 2142091	A	8		

Abstract (Basic): GB 2142091 A

In an active sound control system allowance is made in a relatively uncomplicated circuit for acoustic coupling between a sound generating system for generating a cancelling sound wave and a detector for sensing a sound wave to be cancelled. Unwanted sound from a source is detected by a microphone and cancelled by sound from a **speaker**.

The microphone is connected to the **speaker** by way of a fixed gain amplifier which has a feedback processing system. The system is such that its transfer function takes account of acoustic feedback between the **speaker** and the microphone in deriving, with the amplifier 9, a signal to drive the **speaker**.

USE - For cancelling unwanted sound.

3/6

Title Terms: SOUND; ATTENUATE; SYSTEM; ACTIVE; CONTROL; TECHNIQUE; CONTROL; INCORPORATE; SIGNAL; PROCESSOR; FEED; SIGNAL; SOUND; DETECT; GENERATOR; NEGATIVE; FEEDBACK

Derwent Class: P86; Q51; W04

International Patent Class (Additional): F01N-001/06; G10K-011/16; H04R-003/04

File Segment: EPI; EngPI

15/5/14 (Item 8 from file: 350)

DIALOG(R) File 350:Derwent WPIX

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001767091

WPI Acc No: 1977-L3606Y/197751

Band pass filter with surface acoustic wave devices - uses LC circuit to suppress triple transit echoes in surface wave receiver output

Patent Assignee: ROCKWELL INT CORP (ROCW)

Number of Countries: 001 Number of Patents: 001

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
US 4063202	A	19771213				197751 B

Priority Applications (No Type Date): US 76683608 A 19760505

Abstract (Basic): US 4063202 A

At least one of several transducers on a piezoelectric substrate is externally short-circuited by a series resonance circuit for suppression of triple transit echoes. The resonance circuit is additionally connected to an amplifier via a high input or output

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impedance.

In multi-transducer systems any two transducers are **prevented** from communicating via **echo** signals by such resonance circuits. In **filter** cascades several resonance circuits may be included in the interstage coupling circuit or between the input and output amplifiers on the one hand and the cascade on the other hand.

Title Terms: BAND; PASS; **FILTER**; SURFACE; ACOUSTIC; WAVE; DEVICE; CIRCUIT; SUPPRESS; TRIPLE; TRANSIT; ECHO; SURFACE; WAVE; RECEIVE; OUTPUT

Derwent Class: U25

International Patent Class (Additional): H03H-009/26; H03H-013/00

File Segment: EPI

15/5/15 (Item 9 from file: 350)

DIALOG(R) File 350:Derwent WPIX

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001459950

WPI Acc No: 1976-C2843X/197610

Coin discriminating appts using coin vibration - has selective oscillator running at natural coin frequency and one-cycle selector

Patent Assignee: MITANI SHOJI KK (MITA-N)

Number of Countries: 001 Number of Patents: 001

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
US 3939953	A	19760224			197610	B

Priority Applications (No Type Date): JP 7368678 A 19730620; JP 7368676 A 19730620; JP 7368677 A 19730620

Abstract (Basic): US 3939953 A

The coin discriminating apparatus contains an oscillator circuit having a **feedback loop**. A mechanical **filter** includes a discriminated coin, a **speaker** to vibrate the coin and a sensor to pick up the vibration of the coin. A one cycle selector takes out one period of the vibration frequency generated at the oscillator circuit. Means are provided for quantizing the output signal of the one cycle selector by clock pulses. Counter means includes a scale of -1000 counter and a decoder for counting the number of clock pulses. A bistable circuit has its state reversed on receipt of an output produced at the decoder when contents of the counter run up to the **lower limit** or the upper limit of a predetermined tolerance.

Title Terms: COIN; APPARATUS; COIN; VIBRATION; SELECT; OSCILLATOR; RUN; NATURAL; COIN; FREQUENCY; ONE; CYCLE; SELECT

Derwent Class: T05

International Patent Class (Additional): G07F-003/02

File Segment: EPI

June 27, 2003

File 348:EUROPEAN PATENTS 1978-2003/Jun W04

(c) 2003 European Patent Office

File 349:PCT FULLTEXT 1979-2002/UB=20030626, UT=20030619

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Set	Items	Description
S1	39503	(AUDIO? OR AMPLIFICATION? OR ACOUSTIC? OR HEARING) (2N) (AID? OR DEVICE? OR APPARATUS? OR APPTS) OR SPEAKER?
S2	311902	FILTER?
S3	67065	(BAND? OR HIGH? OR LOW?) () (PASS OR STOP? OR LIMIT?) OR PAS-SBAND?
S4	18783	(FEEDBACK? OR ECHO?) (3N) (LOOP? OR CANCEL? OR PREVENT?)
S5	222	S1(3N)S4
S6	3	S5(S)S2(S)S3
S7	1227	S2(5N)S4
S8	9	S7(S)S1(S)S3
S9	8	S8 NOT S6
S10	7166	S2(3N)ADAPT?
S11	5	S10(S)S1(S)S3(S)S4
S12	2	S11 NOT (S9 OR S6)
S13	39	S1(S)S2(S)S3(S)S4
S14	5	S13 AND IC=H04R-025/00
S15	1	S14 NOT (S12 OR S9 OR S6)

June 27, 2003

6/5,K/1 (Item 1 from file: 348)
DIALOG(R)File 348:EUROPEAN PATENTS
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00782736

HEARING AID

HORHILFSGERAT

PROTHESE AUDITIVE

PATENT ASSIGNEE:

TOPHOLM & WESTERMANN APS, (374270), Ny Vestergaardsvej 25, DK-3500
Vaerloese, (DK), (applicant designated states: AT;CH;DE;DK;IT;LI;NL)

INVENTOR:

ANDERSEN, Henning, Haugaard, Adalsvej 40, DK-2970 Horsholm, (DK)

LEGAL REPRESENTATIVE:

Bohmer, Hans Erich, Dipl.-Ing. (2312), Keplerstrasse 23, 71134 Aidlingen,
(DE)

PATENT (CC, No, Kind, Date): EP 793897 A1 970910 (Basic)
EP 793897 B1 980513
WO 9617493 960606

APPLICATION (CC, No, Date): EP 95921771 950529; WO 95EP2033 950529

PRIORITY (CC, No, Date): DE 4441996 941126

DESIGNATED STATES: AT; CH; DE; DK; IT; LI; NL

INTERNATIONAL PATENT CLASS: H04R-025/00;

NOTE:

No A-document published by EPO

LEGAL STATUS (Type, Pub Date, Kind, Text):

Application: 960911 A International application (Art. 158(1))

Application: 970910 A1 Published application (A1with Search Report
;A2without Search Report)

Examination: 970910 A1 Date of filing of request for examination:
970502

Examination: 971112 A1 Date of despatch of first examination report:
971001

Grant: 980513 B1 Granted patent

Oppn None: 990506 B1 No opposition filed

LANGUAGE (Publication, Procedural, Application): German; German; German

FULLTEXT AVAILABILITY:

Available Text	Language	Update	Word Count
CLAIMS B	(English)	9820	324
CLAIMS B	(German)	9820	271
CLAIMS B	(French)	9820	326
SPEC B	(German)	9820	1722
Total word count - document A			0
Total word count - document B			2643
Total word count - documents A + B			2643

...CLAIMS 3), essentially consists of a subtraction stage (5) with a positive and negative input, a **low - pass filter** (6) and a comparator circuit (7) with holding network controlled by a clock pulse generator...

...the subtraction stage to the output of the comparator stage (7) by way of a **feedback loop**.

3. Hearing aid in accordance with claim 1, characterized in that the clock frequency of the clock pulses...

6/5,K/2 (Item 2 from file: 348)

DIALOG(R)File 348:EUROPEAN PATENTS

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00271449

Superregenerative detector.

Pendelruckkopplungs-Detektor.

June 27, 2003

Detecteur a super-reaction.

PATENT ASSIGNEE:

R.F. MONOLITHICS, INC., (327971), 4441 Sigma Road, Dallas Texas 75244,
(US), (applicant designated states: DE;FR;GB;IT;NL)

INVENTOR:

ASH, Darrell L., 1707 Cartwright Dr. Sachse, Texas 75040, (US)

LEGAL REPRESENTATIVE:

Rackham, Stephen Neil et al (35061), GILL JENNINGS & EVERY, Broadgate
House, 7 Eldon Street, London EC2M 7LH, (GB)

PATENT (CC, No, Kind, Date): EP 271190 A2 880615 (Basic)
EP 271190 A3 890531
EP 271190 B1 940302

APPLICATION (CC, No, Date): EP 87308927 871008;

PRIORITY (CC, No, Date): US 939527 861208

DESIGNATED STATES: DE; FR; GB; IT; NL

INTERNATIONAL PATENT CLASS: H03D-011/04;

CITED PATENTS (EP A): EP 184508 A; US 3405364 A; US 3119065 A; US 4143324 A
; FR 2209255 A

CITED REFERENCES (EP A):

IEEE TRANSACTIONS ON CONSUMER ELECTRONICS, vol. CE-33, no. 3, August
1987, pages 395-404, New York, US; D.L. ASH: "A low cost
superregenerative saw stabilized receiver"

IDEML

ABSTRACT EP 271190 A2

A superregenerative detector utilizing a single transistor and having a surface acoustic wave device in the feed back loop coupling the output to the input to cause oscillation wherein the surface acoustic wave device is a low loss delay line formed as a single phase unidirectional transducer on a quartz substrate.

ABSTRACT WORD COUNT: 55

LEGAL STATUS (Type, Pub Date, Kind, Text):

Application: 880615 A2 Published application (Alwith Search Report
;A2without Search Report)

Search Report: 890531 A3 Separate publication of the European or
International search report

Examination: 891220 A2 Date of filing of request for examination:
891023

Examination: 911016 A2 Date of despatch of first examination report:
910902

Grant: 940302 B1 Granted patent

Oppn None: 950222 B1 No opposition filed

LANGUAGE (Publication,Procedural,Application): English; English; English

FULLTEXT AVAILABILITY:

Available Text	Language	Update	Word Count
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CLAIMS B	(English)	EPBBF1	147
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CLAIMS B	(German)	EPBBF1	142
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CLAIMS B	(French)	EPBBF1	174
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SPEC B	(English)	EPBBF1	2492
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Total word count - document A			0
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Total word count - document B			2955
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Total word count - documents A + B			2955
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...SPECIFICATION switching the RF oscillator between an oscillating and a non-oscillating condition; characterised by a **surface acoustic wave** delay line device in the **feedback loop**, the **device** being a **single** phase **unidirectional** transducer formed on a piezoelectric substrate with electrodes a quarter of a wavelength wide, and...

...there are means for coupling a modulated RF signal to the oscillator input; and a **low pass filter** means coupled to the output to recover the modulation signal.

In the accompanying drawings:-

FIG...

6/5,K/3 (Item 1 from file: 349)
DIALOG(R) File 349:PCT FULLTEXT
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00556232 **Image available**

BAND-LIMITED ADAPTIVE FEEDBACK CANCELLER FOR HEARING AIDS
DISPOSITIF ADAPTATIF DE SUPPRESSION DE L'EFFET LARSEN A BANDE LIMITEE
DESTINE AUX PROTHESES AUDITIVES

Patent Applicant/Assignee:
HOUSE EAR INSTITUTE,

Inventor(s):

GAO Shawn,
SOLI Sigfrid,
CHI Hsiang-Feng,

Patent and Priority Information (Country, Number, Date):

Patent: WO 200019605 A2 20000406 (WO 0019605)
Application: WO 99US22757 19990930 (PCT/WO US9922757)
Priority Application: US 98102557 19980930

Designated States: AE AL AM AT AU AZ BA BB BG BR BY CA CH CN CR CU CZ CZ DE
DE DK DM EE ES FI FI GB GD GE GH GM HR HU ID IL IN IS JP KE KG KP
KR KZ LC LK LR LS LT LU LV MD MG MK MN MW MX NO NZ PL PT RO RU SD SE SG
SI SK SL TJ TM TR TT TZ UA UG UZ VN YU ZA ZW GH GM KE LS MW SD SL SZ
TZ UG ZW AM AZ BY KG KZ MD RU TJ TM AT BE CH CY DE DK ES FI FR GB GR IE
IT LU MC NL PT SE BF BJ CF CG CI CM GA GN GW ML MR NE SN TD TG

Main International Patent Class: H04R-025/00

International Patent Class: H04R-003/02

Publication Language: English

Fulltext Availability:

Detailed Description
Claims

Fulltext Word Count: 9255

English Abstract

An improved method for adaptively cancelling acoustic feedback in hearing aids and other audio amplification devices. Feedback cancellation is limited to a frequency band that encompasses all unstable frequencies. By limiting the bandwidth of the feedback cancellation signal, the distortion due to the adaptive filter is minimized and limited only to the unstable feedback regions. A relatively simple signal processing algorithm is used to produce highly effective results with minimal signal distortion.

French Abstract

L'invention concerne un procede ameliore pour supprimer de maniere adaptative l'effet Larsen dans les protheses auditives et dans d'autres dispositifs audio amplifies. La suppression de l'effet Larsen est limitee a la bande de frequences qui englobe toutes les frequences instables. En limitant la bande de frequences du signal d'annulation de l'effet Larsen, on arrive a reduire au minimum la distorsion provoquée par le filtre adaptatif, qui est limitee uniquement aux zones instables de l'effet Larsen. On utilise un algorithme relativement simple de traitement des signaux pour obtenir des resultats probants, et ce avec une distorsion minimale des signaux.

Fulltext Availability:

Claims

Claim

I A **feedback canceller** for an **audio amplification device** comprising:
an **adaptive filter** ;
means for combining an output of the **adaptive filter** with an input of the **audio amplification device**;

June 27, 2003

a first **band limiting filter** having an input coupled to an output of the audio amplification device and an output coupled to an input of the adaptive **filter**, wherein the first **band limiting filter** has a **passband** limited to a frequency band containing unstable frequencies.
2 The device of claim 1 wherein...

...the output of the audio amplification device to the input thereof.

25 A method for **cancelling feedback** in an **audio amplification device** comprising the steps of applying an output of the audio amplification device to a first **band limiting filter** having a **passband** limited to a frequency band containing unstable frequencies. applying an output of the first **band limiting filter** to an adaptive **filter**; combining an output of the adaptive **filter** with an input of the audio amplification device.

26 The method of claim 25 wherein...path from the output of the audio amplification device to the input thereof

48 A **feedback canceller** for an **audio amplification device** comprising:
means for creating a first delay having an input coupled to an audio output of a hearing aid circuit and an output, &
a first **band limiting filter** having an input coupled to the output of
the first delay means and an output;
an adaptive **filter** having an input coupled to the output of the first
band limiting filter and an output;
means for creating a second delay having an input coupled to a...

...of the second delay means, an inverting input coupled to the output of the adaptive **filter** and an output coupled to the input of the hearing aid processing module;
a second **band limiting filter** having an input coupled to the input of
the second delay means and an output...

...second summing node having a non-inverting input coupled to the output of the second **band limiting filter**, an inverting input coupled to the output of the adaptive **filter** and an output;
means for selecting a **filter coefficient** having a first input coupled to the output of the first **band limiting filter**, a second input coupled to the output of the second summing node and an output for supplying the **filter coefficient** to the adaptive **filter**;
wherein the first and second **band limiting filters** have **passbands** limited to a frequency band containing unstable frequencies.

49 A **feedback canceller** circuit for an **audio amplification device** comprising:
means for creating a delay having an input coupled to an audio output of a hearing aid circuit and an output;
a first **band limiting filter** having an input coupled to the output of
the delay means and an output;
an adaptive **filter** having an input coupled to the output of the first

June 27, 2003

band limiting filter and an output;
a summing node having a non-inverting input coupled to a
conditioned...

...of a hearing aid microphone, an inverting input coupled to the output of
the adaptive filter and an output coupled to the input of
the hearing aid processing module;
a second band limiting filter having an input coupled to the output
of the summing node and an output;
a third band limiting filter having an input coupled to the output
of
the first band limiting filter and an output;
means for selecting a filter coefficient having a first input coupled
to the output of the second band limiting filter and a second input
coupled to the output of the third band limiting filter and an
output for supplying
the filter coefficient to the adaptive filter;
wherein the first, second and third band limiting filters have
passbands limited to a frequency band containing unstable frequencies.

50 The device of claim 49 wherein...

June 27, 2003

9/5,K/1 (Item 1 from file: 348)
DIALOG(R)File 348:EUROPEAN PATENTS
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01534470

ECHO PROCESSING APPARATUS
VORRICHTUNG ZUR ECHOVERARBEITUNG
APPAREIL DE TRAITEMENT D'ECHOS

PATENT ASSIGNEE:

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MATSUOKA, Bunkei, Mitsubishi Denki K.K., 2-3, Marunouchi 2-chome,
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KAJIYAMA, Ikuo, Mitsubishi Denki Engineering K.K., 6-2, Otemachi 2-chome,
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LEGAL REPRESENTATIVE:

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PATENT (CC, No, Kind, Date): EP 1300963 A1 030409 (Basic)
WO 2002095975 021128

APPLICATION (CC, No, Date): EP 2002771733 020520; WO 2002JP4860 020520

PRIORITY (CC, No, Date): JP 2001152888 010522

DESIGNATED STATES: AT; BE; CH; CY; DE; DK; ES; FI; FR; GB; GR; IE; IT; LI;
LU; MC; NL; PT; SE; TR

EXTENDED DESIGNATED STATES: AL; LT; LV; MK; RO; SI

INTERNATIONAL PATENT CLASS: H04B-003/23; H04M-001/60; H04R-003/02;
G10L-021/02

ABSTRACT EP 1300963 A1

An echo canceller has a high - pass filter 11, a speaker 3, an A/D converter 6, and an echo canceller 7. The high - pass filter 11 suppresses a low-frequency component in a received signal. The speaker 3 outputs an acoustic sound of a low-frequency suppressed received signal passed through the high - pass filter 11. The A/D converter 6 converts an acoustic echo from a microphone 4 into a transmission signal of a digital form. The echo canceller 7 generates a pseudo echo signal based on the low-frequency suppressed received signal passed through the high - pass filter 11, and eliminates the acoustic echo to be inputted to the microphone 4 from the speaker 3 by subtracting the pseudo echo signal from the digital signal converted by the A/D converter 6.

ABSTRACT WORD COUNT: 122

NOTE:

Figure number on first page: 1

LEGAL STATUS (Type, Pub Date, Kind, Text):

Application: 030122 A1 International application. (Art. 158(1))

Application: 030122 A1 International application entering European phase.

Application: 030409 A1 Published application with search report

Examination: 030409 A1 Date of request for examination: 20030110

LANGUAGE (Publication, Procedural, Application): English; English; Japanese

FULLTEXT AVAILABILITY:

Available Text	Language	Update	Word Count
CLAIMS A	(English)	200315	692
SPEC A	(English)	200315	7925
Total word count - document A			8617
Total word count - document B			0
Total word count - documents A + B			8617

...ABSTRACT A1

An echo canceller has a high - pass filter 11, a speaker 3,

an A/D converter 6, and an **echo canceller** 7. The **high - pass filter** 11 suppresses a low-frequency component in a received signal.. The **speaker** 3 outputs an acoustic sound of a low-frequency suppressed received signal passed through the **high - pass filter** 11. The A/D converter 6 converts an acoustic echo from a microphone 4 into a transmission signal...

...a pseudo echo signal based on the low-frequency suppressed received signal passed through the **high - pass filter** 11, and eliminates the acoustic echo to be inputted to the microphone 4 from the **speaker** 3 by subtracting the pseudo echo signal from the digital signal converted by the A...

...SPECIFICATION processor having a high-pass filter, aD/A converter, a speaker, a microphone, anA/D converter, and an **echo canceller** . The **high - pass filter** suppress a low-frequency component in a received signal in digital form. The D/A converter converts the low-frequency component passed through the **high - pass filter** to a sound signal. The **speaker** outputs an acoustic based on the sound signal. The microphone has a possibility to input an acoustic echo outputted from the **speaker** . The A/D converter converts the sound signal outputted from the microphone to a digital...

...a pseudo echo signal based on a low-frequency suppressed received signal obtained through the **high - pass filter** and generates a transmission signal by subtracting the pseudo echo signal from the digital...

...suppressed received signal to be inputted to the echo canceller is suppressed together by the **high - pass filter**, it is possible to prevent a deterioration of a calculation accuracy of an adaptive **filter** coefficient in the **echo canceller** , to set the difference between the pseudo echo signal and the echo signal to a...

...to another aspect of the present invention, there is provided an echo processor having a **high - pass filter**, a D/A converter, a speaker, a microphone, an A/D converter, an echo canceller, and a double-talk detector. The **high - pass filter** suppress a low-frequency component in a received signal in digital form. The D/A converter converts the low-frequency component passed through the **high - pass filter** to a sound signal. The speaker outputs an acoustic based on the sound signal
...

...a pseudo echo signal based on a low-frequency suppressed received signal obtained through the **high - pass filter** and generates a transmission signal by subtracting the pseudo echo signal from the digital...

...on the low-frequency component, and controls to halt and start the update of a **filter** coefficient of the **echo canceller** .

It is thereby possible to reduce a non-linear distortion outputted from the speaker. Further...

...suppressed received signal to be inputted to the echo canceller is suppressed together by the **high - pass filter**, it is possible to prevent a deterioration of a calculation accuracy of an adaptive **filter** coefficient in the **echo canceller** , to set the difference between the pseudo echo signal and the echo signal to a...

June 27, 2003

Appareil de communication moins libres avec annuleur d'echo

PATENT ASSIGNEE:

MITSUBISHI DENKI KABUSHIKI KAISHA, (208580), 2-3, Marunouchi 2-chome
Chiyoda-ku, Tokyo 100, (JP), (applicant designated states: DE;FR;GB)

INVENTOR:

Higuchi, Koji, c/o Mitsubishi Denki K.K., Tsushinki Seisakusho, 1-1
Tsukaguchi-honmachi, 8-chome, Amagasaki-shi, Hyogo 661, (JP)
Shiono, Takashi, c/o Mitsubishi Denki K.K., Tsushinki Seisakusho, 1-1
Tsukaguchi-honmachi, 8-chome, Amagasaki-shi, Hyogo 661, (JP)

LEGAL REPRESENTATIVE:

Sajda, Wolf E., Dipl.-Phys. et al (9956), MEISSNER, BOLTE & PARTNER
Postfach 86 06 24, D-81633 Munchen, (DE)

PATENT (CC, No, Kind, Date): EP 692903 A1 960117 (Basic)

APPLICATION (CC, No, Date): EP 95110937 950712;

PRIORITY (CC, No, Date): JP 94161522 940713

DESIGNATED STATES: DE; FR; GB

INTERNATIONAL PATENT CLASS: H04M-009/08;

ABSTRACT EP 692903 A1

A hands-free speaking device is provided which is capable of removing ambient noise at the near side from the speech transmitting signal.

The hands-free speaking device comprises a speaker (51) for reproducing a received speech signal, a microphone (52) for outputting a speech transmitting signal, an echo canceler (55) for removing an echo (62) transmitted from the speaker (51) to the microphone (52), and a voice detector (11) comprising a noise level monitoring circuit (2) for monitoring the level of the ambient noise (63) input to the microphone (52), a voice level monitoring circuit (3) for monitoring the level of a voice signal (61) input to the microphone (52), and a comparator (4) for generating a signal for increasing the attenuation amount of an attenuator (10) to attenuate the transmitting speech signal when the noise level is higher than the voice level. (see image in original document)

ABSTRACT WORD COUNT: 170

LEGAL STATUS (Type, Pub Date, Kind, Text):

Examination: 020327 A1 Date of dispatch of the first examination report: 20020206

Application: 960117 A1 Published application (A1with Search Report ;A2without Search Report)

Examination: 960327 A1 Date of filing of request for examination: 960124

LANGUAGE (Publication, Procedural, Application): English; English; English

FULLTEXT AVAILABILITY:

Available Text	Language	Update	Word Count
CLAIMS A	(English)	EPAB96	537
SPEC A	(English)	EPAB96	3695
Total word count - document A			4232
Total word count - document B			0
Total word count - documents A + B			4232

...SPECIFICATION 54, an acoustic echo canceler 55, and a 2-line/4-line converter 57.

The **high - pass** filter 12 attenuates the low-frequency component in the speech transmitting signal generated from the speech transmitting signal microphone 52. The high-frequency component alone is passed through the **filter** and output to the acoustic **echo canceler** 55. The **low - pass** **filter** 13 attenuates the high-frequency component in the output of the acoustic echo canceler 55...

...to the attenuator 10. The attenuator 10, the voice detector 11, the received speech signal **speaker** 51, the speech transmitting signal microphone 52, the speech transmitting side output terminal 53, the...

June 27, 2003

DIALOG(R) File 348:EUROPEAN PATENTS
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00252199

Programmable sound reproducing system.

Programmierbares Schallwiedergabesystem.

Système de reproduction sonore programmable.

PATENT ASSIGNEE:

AUDIMAX CORPORATION, (1522870), c/o Energy Transportation Group, Inc.
1185 Avenue of the Americas, New York, New York 10036, (US), (applicant
designated states: AT;BE;CH;DE;ES;FR;GB;GR;IT;LI;LU;NL;SE)

INVENTOR:

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LEGAL REPRESENTATIVE:

Rackham, Stephen Neil (35061), GILL JENNINGS & EVERY, Broadgate House, 7
Eldon Street, London EC2M 7LH, (GB)

PATENT (CC, No, Kind, Date): EP 250679 A2 880107 (Basic)
EP 250679 A3 890419
EP 250679 B1 930707

APPLICATION (CC, No, Date): EP 86307717 861007;

PRIORITY (CC, No, Date): US 879214 860626

DESIGNATED STATES: AT; BE; CH; DE; ES; FR; GB; GR; IT; LI; LU; NL; SE

INTERNATIONAL PATENT CLASS: H04R-025/00; H03G-005/16; H04R-003/02;

CITED PATENTS (EP A): US 4188667 A; EP 71845 A; GB 1582821 A; US 4131760 A;
NL 8500595 A

ABSTRACT EP 250679 A2

In a hearing aid system, selected optimum parameter values are programmed into an electronically erasable, programmable read-only memory (EEPROM) (84) which supplies coefficients to a programmable filter (64) and amplitude limiter (67) in the hearing aid so as to cause the hearing aid to adjust automatically to the optimum set of parameter values for the speech level, room reverberation, and type of background noise then obtaining. The programmable filter may be a digital equivalent of a tapped delay line in which each delayed sample is multiplied by a weighting coefficient, and the sum of the weighted samples generates a desired electro-acoustic characteristic; or a tapped analog delay line in which the sum of the weighted outputs of the taps generates the desired characteristics.

Acoustical feedback is reduced by an electrical feedback path in the hearing aid which is matched in both amplitude and phase to the acoustic feedback path, the two feedback signals being subtracted so as to cancel each other. Alternatively, a single filter in the forward path may be used for this purpose with a transmission characteristic equivalent to that of the programmable filter in the forward path plus the electrical feedback path. Also, the relative speech-noise content in the signals from the hearing aid microphone is sensed and binary words are generated and supplied to the programmable filter for selecting from memory a set of delay line tap coefficients that are effective to impart to the filter the appropriate frequency response for the specific environmental noise condition being detected.

ABSTRACT WORD COUNT: 256

LEGAL STATUS (Type, Pub Date, Kind, Text):

Lapse: 20000126 B1 Date of lapse of European Patent in a
contracting state (Country, date): GR
19930707, LU 19931031,

Application: 880107 A2 Published application (A1with Search Report
;A2without Search Report)

Change: 880831 A2 Representative (change)

Search Report: 890419 A3 Separate publication of the European or
International search report

Examination: 891129 A2 Date of filing of request for examination:

June 27, 2003

891004
Examination: 910807 A2 Date of despatch of first examination report:
910620
Change: 920902 A2 Representative (change)
*Assignee: 920902 A2 Applicant (transfer of rights) (change):
AUDIMAX CORPORATION (1522870) c/o Energy
Transportation Group, Inc. 1185 Avenue of the
Americas New York, New York 10036 (US)
(applicant designated states:
AT;BE;CH;DE;ES;FR;GB;GR;IT;LI;LU;NL;SE)
Grant: 930707 B1 Granted patent
Oppn None: 940629 B1 No opposition filed
Lapse: 991229 B1 Date of lapse of European Patent in a
contracting state (Country, date): LU
19931031,

LANGUAGE (Publication, Procedural, Application): English; English; English

FULLTEXT AVAILABILITY:

Available Text	Language	Update	Word Count
CLAIMS B	(English)	EPBBF1	1366
CLAIMS B	(German)	EPBBF1	1487
CLAIMS B	(French)	EPBBF1	1773
SPEC B	(English)	EPBBF1	6717
Total word count - document A			0
Total word count - document B			11343
Total word count - documents A + B			11343

...SPECIFICATION aid, switching back and forth two sets of electroacoustic characteristics at will by means of the switches 123A and 123B, choosing the characteristic which is more intelligible or preferable in some way. Paired comparisons made...

...88 is connected to the switch 37, the counter 74 and the RAM 71.

The hearing aid is now in its normal operating mode and speech detected by the microphone 57 is...

...gain control circuit 58 and transmitted through the amplifier 60, the filter 63 and the low pass filter 63a into the programmable filter circuitry.

A so-called "bucket brigade" analog delay time...

9/5,K/4 (Item 1 from file: 349)

DIALOG(R) File 349:PCT FULLTEXT

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00776439 **Image available**

DC OFFSET CALIBRATION FOR A DIGITAL SWITCHING AMPLIFIER
CALIBRAGE DE DECALAGE EN CONTINU POUR AMPLIFICATEUR DE COMMUTATION
NUMERIQUE

Patent Applicant/Assignee:

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Patent and Priority Information (Country, Number, Date):

Patent: WO 200110012 A1 20010208 (WO 0110012)

Application: WO 2000US20197 20000725 (PCT/WO US0020197)

Priority Application: US 99146416 19990729; US 2000624503 20000724

Designated States: AE AG AL AM AT AU AZ BA BB BG BR BY BZ CA CH CN CR CU CZ
DE DK DM DZ EE ES FI GB GD GE GH GM HR HU ID IL IN IS JP KE KG KP KR KZ
LC LK LR LS LT LU LV MA MD MG MK MN MW MX MZ NO NZ PL PT RO RU SD SE SG
SI SK SL TJ TM TR TT TZ UA UG UZ VN YU ZA ZW

June 27, 2003

(EP) AT BE CH CY DE DK ES FI FR GB GR IE IT LU MC NL PT SE
(OA) BF BJ CF CG CI CM GA GN GW ML MR NE SN TD TG
(AP) GH GM KE LS MW MZ SD SL SZ TZ UG ZW
(EA) AM AZ BY KG KZ MD RU TJ TM

Main International Patent Class: H03F-001/02

International Patent Class: H03F-003/217; G01R-019/00

Publication Language: English

Filing Language: English

Fulltext Availability:

Detailed Description

Claims

Fulltext Word Count: 4594

English Abstract

An offset voltage calibration circuit for use with a digital switching amplifier (400). The calibration circuit includes an analog-to-digital converter (406) for converting at least one DC offset voltage associated with the digital switching amplifier (400) to digital offset data. A memory (408) stores the digital offset data. Control circuitry (402) controls the analog-to-digital converter (406). A digital-to-analog converter (404) coupled to the memory (408) receives the digital offset data and generates an offset compensation voltage for applying to an input port of the digital switching amplifier which thereby cancels at least a portion of the at least one DC offset voltage.

French Abstract

L'invention concerne un circuit de calibrage de tension de decalage a utiliser avec un amplificateur (400) de commutation numerique. Ledit circuit de calibrage comprend un convertisseur analogique-numerique (406) permettant de convertir au moins une tension de decalage en continu associee a l'amplificateur (400) de commutation numerique en donnees de decalage numerique. Une memoire (408) stocke les donnees de decalage numerique. Des circuits de commande (402) commandent le convertisseur analogique-numerique (406). Un convertisseur numerique-analogique (404) relie a la memoire (408) reçoit les donnees de decalage numerique et genere une tension de compensation de decalage a appliquer a une borne d'entree de l'amplificateur de commutation numerique qui annule ainsi au moins une partie d'une tension de decalage en continu au minimum.

Legal Status (Type, Date, Text)

Publication 20010208 A1 With international search report.

Examination 20010614 Request for preliminary examination prior to end of 19th month from priority date

Fulltext Availability:

Detailed Description

Detailed Description

... amplifier I 00 and converted to a one-bit signal by a noise-shaping oversampled **feedback loop** which includes **loop filter** 102, comparator 104, break-before-make generator 106, power stage driver 108, and power stage...

...The one-bit signal drives the power stage I 10 which, in turn, drives a **low pass** filter comprising inductor I 12 and capacitor I 14 which recovers the audio signal with which **speaker** I 16 is driven.

Any DC offset inherent in amplifier I 00 is amplified...

9/5,K/5 (Item 2 from file: 349)

DIALOG(R)File 349:PCT FULLTEXT

(c) 2003 WIPO/Univentio. All rts. reserv.

00749056 **Image available**

GATEWAY WITH VOICE

PASSERELLE VOCALE

Patent Applicant/Assignee:

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(Residence), US (Nationality), (For all designated states except: US)

Patent Applicant/Inventor:

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(Residence), US (Nationality), (Designated only for: US)

Legal Representative:

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Pasadena, CA 91109-7068, US,

Patent and Priority Information (Country, Number, Date):

Patent: WO 200062501 A2-A3 20001019 (WO 0062501)
Application: WO 2000US10149 20000413 (PCT/WO US0010149)
Priority Application: US 99129134 19990413; US 99136685 19990528; US
99154903 19990920; US 99156266 19990927; US 99157470 19991001; US
99160124 19991018; US 99161152 19991022; US 99162315 19991028; US
99163169 19991102; US 99163170 19991102; US 99163600 19991104; US
99164379 19991109; US 99164689 19991110; US 99164690 19991110; US
99166289 19991118; US 99454219 19991209; US 99171203 19991215; US
99171169 19991216; US 99171180 19991216; US 99171184 19991216; US
2000178258 20000125; US 2000493458 20000128; US 2000522185 20000309

Designated States: AE AL AM AT AU AZ BA BB BG BR BY CA CH CN CR CU CZ DE DK
DM EE ES FI GB GD GE GH GM HR HU ID IL IN IS JP KE KG KP KR KZ LC LK LR
LS LT LU LV MA MD MG MK MN MW MX NO NZ PL PT RO RU SD SE SG SI SK SL TJ
TM TR TT TZ UA UG US UZ VN YU ZA ZW
(EP) AT BE CH CY DE DK ES FI FR GB GR IE IT LU MC NL PT SE
(OA) BF BJ CF CG CI CM GA GN GW ML MR NE SN TD TG
(AP) GH GM KE LS MW SD SL SZ TZ UG ZW
(EA) AM AZ BY KG KZ MD RU TJ TM

Main International Patent Class: H04L-012/28

International Patent Class: H04L-012/66; H04L-007/02; H04B-003/23

Publication Language: English

Filing Language: English

Fulltext Availability:

Detailed Description

Claims

Fulltext Word Count: 80268

English Abstract

In one aspect of the present invention, a network gateway is configured to facilitate on line and off line bi-directional communication between a number of near end data and telephony devices with far end data termination devices via a hybrid fiber coaxial network and a cable modem termination system. The described network gateway combines a QAM receiver, a transmitter, a DOCSIS MAC, a CPU, a voice and audio processor, an Ethernet MAC, and a USB controller to provide high performance and robust operation.

French Abstract

Selon un aspect de la presente invention, une passerelle de reseau est conçue pour faciliter la communication bidirectionnelle en-ligne et hors-ligne entre, d'une part, une pluralite de dispositifs de telephonie et de traitement de donnees d'extremite rapprochee et, d'autre part, des dispositifs terminaux de traitement de donnees d'extremite eloignee, par l'intermediaire d'un reseau a systeme de transmission hybride fibre et coaxial et d'un systeme de terminaison a modem cable. La passerelle de reseau de cette invention combine un recepteur QAM, un emetteur, un MAC DOCSIS, une unite centrale, un processeur de donnees vocales et sonores, un MAC Ethernet et un controleur USB dans le but d'assurer de hautes performances et un fonctionnement robuste.

Legal Status (Type, Date, Text)

Publication 20001019 A2 Without international search report and to be republished upon receipt of that report.
Examination 20010412 Request for preliminary examination prior to end of 19th month from priority date
Search Rpt 20020103 Late publication of international search report
Republication 20020103 A3 With international search report.

Fulltext Availability:

Detailed Description

Detailed Description

... embodiment of the present invention;

FIG. 18A is a block diagram of a single pole **low pass** filter used to smooth or average the differences between sampling rates in accordance with a...

...typically associated with conventional echo cancellers and utilizes the delay associated with a decimator and **high pass** filter to provide a look ahead capability so that filter I 0 adaptation may be...

...the present invention;

I 0 FIG. 31 is a block diagram of a method for **cancelling** echo returns in accordance with a

9/5,K/6 (Item 3 from file: 349)

DIALOG(R) File 349:PCT FULLTEXT

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00514343 **Image available**

METHOD AND APPARATUS FOR MONITORING, CONTROLLING, AND CONFIGURING REMOTE COMMUNICATION DEVICES

PROCEDE ET APPAREIL DE CONTROLE, DE COMMANDE ET DE CONFIGURATION DE DISPOSITIFS DE COMMUNICATION DISTANTS

Patent Applicant/Assignee:

CONEXANT SYSTEMS INC,

Inventor(s):

COLLIN Zeev,
TAMIR Tal,

June 27, 2003

Patent and Priority Information (Country, Number, Date):

Patent: WO 9945695 A1 19990910
Application: WO 99US4841 19990304 (PCT/WO US9904841)
Priority Application: US 9876784 19980304; US 98154643 19980917; US
98193304 19981117

Designated States: AL AM AU AZ BA BB BG BR BY CA CN CU CZ EE GE GH GM HR HU
ID IL IS JP KE KG KP KR KZ LC LK LR LS LT LV MD MG MK MN MW MX NO NZ PL
RO RU SD SG SI SK SL TJ TM TR TT UA UG UZ VN YU ZW AT BE CH CY DE DK ES
FI FR GB GR IE IT LU MC NL PT SE

Main International Patent Class: H04M-011/06

International Patent Class: H04L-012/26; H04L-012/24

Publication Language: English

Fulltext Availability:

Detailed Description

Claims

Fulltext Word Count: 8571

English Abstract

A communication system for monitoring and/or controlling communication parameters of a remote communication device. The communication system monitors a communication channel that is created between the remote communication device and controls the communication device by adjusting internal settings of the communication device that represent communication parameters. The communication device is communicatively coupled to a communication channel to carry out ongoing communications between the communication device and the communication channel. Further, a software module is associated with the communication device, and the software module accesses the internal settings of the communication device from a remote location via the communication channel and performs diagnostics such as monitoring, controlling, and configuring the communication device using the internal settings of the communication device.

French Abstract

Système de communication destiné à contrôler et/ou commander les paramètres de communication d'un dispositif de communication distant. Le système de communication contrôle une voie de communication créée entre le dispositif de communication distant et il commande le dispositif de communication en ajustant les réglages internes du dispositif de communication représentant les paramètres de communication. Le dispositif de communication est couplé de manière communicative à une voie de communication afin de permettre des communications permanentes entre le dispositif de communication et la voie de communication. De plus, un module logiciel est associé au dispositif de communication, le module logiciel accède au réglage interne du dispositif de communication depuis un point distant par la voie de communication et il exécute un diagnostic tel que le contrôle, la commande et la configuration du dispositif de communication au moyen des réglages internes de ce dernier.

Fulltext Availability:

Claims

Claim

... 0 indicates
u-law
RKCFG -ENCODING-LAW V90-RK-CODES, BOOL
(TRUE=A-Law) None
***** SpeakerPhone Constants
Hardware Delay
RKCFG -EC-DELAY SPKP-RK-CODES, //@SPKP
MODULE, INT No of
Samples...

...RKMON DATA RES ECHO GET
RKCTL-DATA-RES-ECHO-REQUEST=v34-RK-CODESI// None
None

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```
SpeakerPhone Constants
Speakerphone Mode (FD, HD, HS)
RKCTL-SPKP-MODE SPKP-RK-CODES, SPKPMODE
None
Output Mute
RKCTL IO MUTE,
@SPKP PROBE,BOOL - Yes/No None
// Echo Cancellers
RKCTL FILTER -LENGTH, @SPKP-MODULE, INT - No
of Taps) None
RKCTL EC-OPERATE, (SPKP
MODULE,BOOL
Yes...bit)
PCM Pad
RKMON-PAD-DETECTED,
None DWORD PAD 0=NORMAL 3=3dBPad 6=6dBPad
// High Pass filter enabled
RKMON-HIGHPASS-FILTER-ENABLED
None BOOL Yes/No
SpeakerPhone Constants
Speakerphone Mode (FD, HD, HS)
RKMON-SPKP-MODE SPKP-RK-CODES, None
so SPKPMODE
State
RKMON...
...Yes/No
RKMON-SATURATION,
SPKP PROBE BOOL - Yes/No
RKMON-DC-LEVEL,
SPKP PROBE FLOAT
Echo Cancellers
RKMON- FILTER -LENGTH,
SPKP MODULE INT No of Taps
RKMON-EC-OPERATEI
SPKP-MODULE BOOL Yes/No...
```

9/5,K/7 (Item 4 from file: 349)
DIALOG(R)File 349:PCT FULLTEXT
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00514342 **Image available**
METHOD AND APPARATUS FOR MONITORING, CONTROLLING, AND CONFIGURING LOCAL
COMMUNICATION DEVICES

PROCEDE ET APPAREIL DE SURVEILLANCE, REGLAGE ET CONFIGURATION DE
DISPOSITIFS LOCAUX DE TRANSMISSION

Patent Applicant/Assignee:
CONEXANT SYSTEMS INC,

Inventor(s):
COLLIN Zeev,
TAMIR Tal,

Patent and Priority Information (Country, Number, Date):

Patent: WO 9945694 A1 19990910

Application: WO 99US4690 19990304 (PCT/WO US9904690)

Priority Application: US 9876784 19980304; US 98154643 19980917; US
98192627 19981116

Designated States: AL AM AU AZ BA BB BG BR BY CA CN CU CZ EE GE GH GM HR HU
ID IL IS JP KE KG KP KR KZ LC LK LR LS LT LV MD MG MK MN MW MX NO NZ PL
RO RU SD SG SI SK SL TJ TM TR TT UA UG UZ VN YU ZW AT BE CH CY DE DK ES
FI FR GB GR IE IT LU MC NL PT SE

Main International Patent Class: H04M-011/06

International Patent Class: H04L-012/26; H04L-012/24

Publication Language: English

Fulltext Availability:

Detailed Description
Claims
Fulltext Word Count: 8979

English Abstract

A communication system for monitoring and/or controlling communication parameters of a communication device. The communication system monitors a communication channel that is created when the communication device connects to a network, controls the communication device as it operates on the network, and configures the communication device. The communication device is commonly a modem and is communicatively coupled to the network to carry out ongoing communications between the modem and the network through the communication channel. Further, a software module is associated with the modem, and the software module accesses the internal settings of the modem via the communication channel (if necessary) and performs operations such as monitoring, controlling, and configuring the modem (or other communication device) using the internal settings of the modem.

French Abstract

L'invention porte sur un système de télécommunications surveillant et/ou régulant les paramètres d'un système de télécommunications. L'édit système surveille un canal de transmission créé au moment où le dispositif de télécommunications se raccorde à un réseau, le commande alors qu'il opère sur le réseau, et le configure. Le dispositif de télécommunications est habituellement un modem raccordé au réseau et acheminant les communications en cours entre le modem et le réseau et transitant par le canal de transmission. En outre, un logiciel associé au modem et ayant accès aux réglages intérieurs du modem (si nécessaire par l'intermédiaire du canal de transmission), exécute des opérations telles que la surveillance, le réglage, ou la configuration du modem (ou d'autres dispositifs de transmission) en utilisant les réglages intérieurs du modem.

Fulltext Availability:

Claims

Claim

... 0 indicates
u-law
RKCFG-ENCODING-LAW V90-RK-CODES, BOOL
(TRUE=A-Law) None
***** **SpeakerPhone Constants**
Hardware Delay
RKCFG-EC- DELAY SPKP-RK-CODES, // (SPKP
MODULE, INT No of
Samples...)

...RKMON DATA RES ECHO GET
RKCTL-DATA-RES-ECHO-REQUEST=v34-RK-CODES, // None
None
SpeakerPhone Constants
Speakerphone Mode (FD, HD, HS)
RKCTL-SPKP-MODE SPKP-RK-CODES, SPKPMode
None
Output Mute
RKCTL-TO-MUTE,
@SPKP-PROBEIBOOL - Yes/No@ None
// **Echo Cancellers**
RKCTL- **FILTER** -LENGTH, @SPKP-MODULE, INT - No
of Taps1 None
RKCTL-EC-OPERATE, @SPKP-MODULE, BOOL
Yes...bit)
PCM Pad
RKMON- PAD-DETECTED,

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```
None DWORD PAD 0=NORMAL 3=3dBPad 6=6dBPad
// High Pass filter enabled
RKMON-HIGHPASS-FILTER-ENABLED
None BOOL Yes/No
SpeakerPhone Constants
Speakerphone Mode (FD, HD, HS)
RKMON-SPKP-MODE SPKP-RK-CODES, None
SPKPMode
State
RKMON-STATE...
...Yes/No
RKMON-SATURATION,
SPKP-PROBE BOOL - Yes/No
RKMON-DC-LEVELI
SPKP-PROBE FLOAT
// Echo Cancellers
RKMON- FILTER -LENGTHI
SPKP MODULE INT No of Taps
RKMON-EC-OPERATE,
SPKP MODULE BOOL Yes/No...
```

9/5,K/8 (Item 5 from file: 349)
DIALOG(R) File 349:PCT FULLTEXT
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00191306

HEARING AID

PROTHESE AUDITIVE

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Inventor(s):

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SVEAN Jarle,
RAMSTAD Tor Audun,

Patent and Priority Information (Country, Number, Date):

Patent: WO 9108654 A1 19910613
Application: WO 90NO178 19901129 (PCT/WO NO9000178)
Priority Application: NO 89486 19891130

Designated States: AT AT AU BB BE BF BG BJ BR CA CF CG CH CH CM DE DE DK DK
ES ES FI FR GA GB GB GR HU IT JP KP KR LK LU LU MC MG ML MR MW NL NL NO
RO SD SE SE SN SU TD TG US

Main International Patent Class: H04R-025/00

Publication Language: English

Fulltext Availability:

Detailed Description
Claims

Fulltext Word Count: 12223

English Abstract

Programmable hybrid hearing aid with digital signal processing comprising a main section (1) which can be inserted in the meatus (6). The main section (1) comprises an open connection between the ear opening and an inner portion of the meatus (6), providing an acoustic transmission channel with low-pass characteristic and resonant amplification. The main section further comprises an electroacoustic transmission channel based on digital signal processing and a signal processor (DSP) and with possibility for suppressing a possible acoustic signal feedback through the acoustic transmission channel. A variant of the hearing aid is provided with a microphone (M1) and the feedback signal is suppressed by digital filtering. Another variant of the hearing

aid employs two microphones (M1, M2), and the feedback signal may then be suppressed by phasing out before the digital signal processing, while the digital signal processing also comprises cancellation of the feedback signal in case of high gain. A number of response functions are stored in a memory (RAM2) in a control unit and is freely chosen by the user in regard of adaption to hearing function and acoustic environment. All the electronics of the electroacoustic channel in the hearing aid is implemented as a monolithic integrated circuit (3) in CMOS technology.

French Abstract

Prothese auditive hybride programmable avec traitement des signaux numeriques. Elle comprend une partie principale (1) a introduire dans le conduit auditif (6). Celle-ci (1) comporte une liaison ouverte entre l'orifice de l'oreille et la partie interne du conduit auditif (6), constituant ainsi un canal de transmission acoustique a effet de passe-bas et d'amplification par resonance. Cette partie principale comprend en outre un canal de transmission electroacoustique base sur le traitement de signaux numeriques, ainsi qu'une unite de traitement des signaux. Il est en outre possible de supprimer une eventuelle reaction parasite acoustique grace au canal de transmission acoustique. Une autre version de cette prothese auditive prevoit un microphone (M1), le signal de reaction etant supprime par filtrage numerique. Dans une troisieme version, on a deux microphones (M1, M2), et le signal de reaction peut alors etre supprime par elimination progressive de phase avant le traitement du signal numerique, qui entraine aussi l'elimination du signal de retour en cas de gain eleve. Un certain nombre de fonctions de reaction sont memorisees (RAM2) dans une unite de commande. L'utilisateur peut librement les selectionner selon leur degré d'adaptation a la fonction auditive et a l'environnement acoustique. Toutes les pieces electroniques du canal electro-acoustique de ladite prothese sont realisees sous forme de circuit integre monolithe (3) par technique CMOS.

Fulltext Availability:

Claims

Claim

achieved with a **hearing aid** which is characterized by the features presented by the characteristic part of claim 5.
A method for detection and signal processing in a **hearing aid** principally of the type presented in claim 5. is characterized by the features presented by the characteristic part of claim 13,
Further features and advantages of the **hearing aid** in accordance with the invention are presented in the appended independent claims 2. 4 and...

...with the attached drawings.

Fig. 1a is a block diagram showing the principles of a **hearing aid** in accordance with the invention.

Fig. 1b is a schematic representation of an electrical equivalent connection for the acoustic channel in fig. 1a.

Fig. 2 is a variant of the **hearing aid** in accordance with the invention.

Fig. 3 is a further variant of the **hearing aid** in accordance with the invention.

Fig. 4a is a schematic block diagram for a **hearing aid** in accordance with the invention, where one microphone is used.

Fig. 4b shows the **hearing aid** in fig. 4a with a cancellation **filter** inserted in a **feedback loop**.

Fig. 4c shows the **hearing aid** in fig. 4a with a cancellation filter inserted in the signal's forward path.

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Fig. 4d shows the **hearing aid** in fig. 4a with a power amplifier in the output stage.

Fig. 5a shows a **hearing aid** in accordance with the invention, where two microphones are used,

Fig. 5b shows a digital signal processor used with the **hearing aid** in fig. 5a.

Fig. 6a is three examples of response curves for strong, moderate and weak hearing impairment respectively, in addition to the sound pressure response of a meatus without **hearing aid**.

Fig. 6b is an example of the response curve for an envelope signal and a...

...in the digital signal processor in fig. 5b, The principles of the design of a **hearing aid** in accordance with the invention are illustrated schematically in fig. 1a. The **hearing aid** comprises an electroacoustic channel consisting of an analog input section, a digital signal processor and...

...analog output section together with an acoustic transmission channel which simultaneously constitutes both an acoustic **low pass** filter and a potential acoustic feedback path, An external sound field is detected by a to the detector via its acoustic channel, The method of construction of the **hearing aid** causes a section of the inner meatus near the tympanum also to constitute an active component of the **hearing aid** by acting as a resonator.

The acoustic channel will be discussed in more detail in connection with the equivalence diagram in fig. 1b..

Fig. 2 shows a variant of the **hearing aid** in accordance with the invention. This variant comprises a main section with an acoustic transmission...

...a distance from the first microphone M1. The electronic components which form part of the **hearing aid** are provided in a first secondary section 2a which here is positioned in the concha...

...this secondary section 2a it may be appropriate to provide a battery 4 for the **hearing aid**. Another not shown secondary section constitutes a case for the **hearing aid**, on an inner end of the main section 1 is provided a miniaturised sound generator SG which faces the tympanum and converts the amplified electrical signal in the **hearing aid** to an **acoustic** signal which is intercepted by the tympanum, In order to have room inside a person...

...must preferably have a diameter which is less than approximately 4,5 mm. In the **hearing aid** in accordance with the present invention an electrodynamic sound generator of the type described in fig. 3 the **hearing aid** in accordance with the invention is shown in a variant with two microphones M1 and...

...to the main section 1 in fig. 2. All the electronics as well as the **hearing aid**'s battery 4 are provided in the main section 1, so that a secondary section provided in or beside the concha has been dispensed with, The **hearing aid**'s main section 1 has rather been connected with a not shown secondary section 2 in the form of a case in which the main section is kept when the **hearing aid** is not in use and which may also comprise possible electronic and electrical auxiliary devices...

June 27, 2003

...possibly plugs and switches and is arranged so that it is used for charging the **hearing aid**'s battery 4 when the main section I is in the case. The main section...

...charging.

The electrical and electronic components used for signal processing in a variant of the **hearing aid** in accordance with the invention with one microphone M1, will now be described in more...

...fig. 4a, All of these components can be provided in a suitable manner in the **hearing aid**'s main section 1 or possibly in a first secondary section 2a, The microphone M1...

...frequency of, e.g., 8 kHz. This will therefore be the upper limit of the **hearing aid**'s frequency response, The microphone M1 may be, e.g., a cardiod micropohone which gives...front of the inputs of a sound generator SG.'In order to eliminate any acoustic **feedback** a **cancellation filter** 35 is used which in fig, 4b is shown inserted in a feedback loop between...

...amplifier 15 are all connected to a battery 4 which is preferably provided in the **hearing aid**'s main section 1.

Fig. 5a shows the electronic components for signal processing in a **hearing aid** in accordance with the invention which uses two microphones M1, M2. In the figure the...

...the input to the first channel CH1 and a second channel CH2 respectively in the **hearing aid**'s analog section, Each channel CH1, CH2 thus comprises a series connection of an impedance...

June 27, 2003

12/5,K/1 (Item 1 from file: 348)
DIALOG(R) File 348:EUROPEAN PATENTS
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00616101

METHOD FOR FAIL-SAFE OPERATION IN A SPEAKER PHONE SYSTEM
AUSFALLGESICHERTES BETRIEBSVERFAHREN IN EINEM LAUTFERNSPRECHSYSTEM
PROCEDE DE FONCTIONNEMENT A SECURITE INTEGREE DANS UN SYSTEME TELEPHONIQUE
A HAUT PARLEUR

PATENT ASSIGNEE:

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PATENT (CC, No, Kind, Date): EP 648397 A1 950419 (Basic)
EP 648397 A1 990512
EP 648397 B1 020918
WO 94001957 940120

APPLICATION (CC, No, Date): EP 93916812 930628; WO 93US6140 930628

PRIORITY (CC, No, Date): US 909060 920702; US 943732 920911

DESIGNATED STATES: DE; GB

INTERNATIONAL PATENT CLASS: H04M-001/00; H04M-009/08; H04B-003/23

CITED PATENTS (EP B): US 4600815 A; US 4629829 A; US 4720856 A; US 4796287
A; US 4845746 A; US 4912758 A; US 4965822 A; US 4984265 A; US 5054061 A

NOTE:

No A-document published by EPO

LEGAL STATUS (Type, Pub Date, Kind, Text):

Change: 011017 A1 International Patent Classification changed:
20010828

Application: 940427 A International application (Art. 158(1))

Grant: 020918 B1 Granted patent

Examination: 020313 A1 Date of dispatch of the first examination
report: 20020128

Assignee: 020807 A1 Transfer of rights to new applicant: Polycom,
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95035-7409 US

Application: 950419 A1 Published application (A1with Search Report.
;A2without Search Report)

Examination: 950419 A1 Date of filing of request for examination:
941229

Search Report: 990512 A1 Drawing up of a supplementary European search
report: 990326

Change: 990512 A1 Obligatory supplementary classification
(change)

LANGUAGE (Publication, Procedural, Application): English; English; English

FULLTEXT AVAILABILITY:

Available Text	Language	Update	Word Count
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CLAIMS B	(English)	200238	138
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CLAIMS B	(German)	200238	133
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CLAIMS B	(French)	200238	180
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SPEC B	(English)	200238	3671
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Total word count - document A			0
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Total word count - document B			4122
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Total word count - documents A + B			4122
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...SPECIFICATION Reference Manual" by AT&T, Oct. 1989.

Speaker signal 45 from telephone lines 44 is low - pass filtered and
digitized at 8 kilohertz by A/D converter 47. The digitized speaker
signal 51 is combined with Line Echo Canceler (LEC) signal 49 in

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summer 53 to produce summer output 55. As in AEC 26, LEC 40 uses an **adaptive filter** to model the impulse response of the sidetones coming from the near-end signals via microphone 12. The replica of the sidetones is then eliminated from the **speaker** signals 51 to prevent the near-end talkers from hearing their own voices coming back. Similar to the **adaptive filter** in AEC 26, a conventional normalized least mean square algorithm is used. Summer output 55 is fed back to LEC 40 to measure the effectiveness of the **echo cancellation**.

Following **echo cancellation** on the receive side, summer output 55 connects to receive attenuator 57, which functions in...

12/5,K/2 (Item 1 from file: 349)

DIALOG(R)File 349:PCT FULLTEXT

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00253802

**METHOD FOR FAIL-SAFE OPERATION IN A SPEAKER PHONE SYSTEM
PROCEDE DE FONCTIONNEMENT A SECURITE INTEGREE DANS UN SYSTEME TELEPHONIQUE
A HAUT PARLEUR**

Patent Applicant/Assignee:

POLYCOM INC,

Inventor(s):

HUANG Shan-Shan,

HINMAN Brian L,

GAUT Eric K,

Patent and Priority Information (Country, Number, Date):

Patent: WO 9401957 A1 19940120

Application: WO 93US6140 19930628 (PCT/WO US9306140)

Priority Application: US 92909060 19920702; US 92943732 19920911

Designated States: CA JP KR AT BE CH DE DK ES FR GB GR IE IT LU MC NL PT SE

Main International Patent Class: H04M-001/00

International Patent Class: H04M-03:00

Publication Language: English

Fulltext Availability:

Detailed Description

Claims

Fulltext Word Count: 4501

English Abstract

A method is disclosed for maintaining loop stability between a transmit and a receive path of a speaker phone which is capable of operating in either a full-duplex or a half-duplex mode. Loop stability is maintained and normal operation is achieved by controlling transmit and receive path attenuators. The method first determines whether feedforward or feedback signals levels should be used (76). Based on this determination, the signal and noise parameters along various points in the transmit and receive path are evaluated (78, 80). Based on the parameter values (82), the transmit and receive attenuators are adjusted to maintain loop stability and to operate the speaker phone in the proper state (84, 75).

French Abstract

L'invention se rapporte à un procédé permettant de maintenir la stabilité de boucle entre une voie de transmission et une voie de réception d'un poste téléphonique à haut-parleur pouvant fonctionner soit en mode duplex intégral soit en mode semi-duplex. La stabilité de boucle est maintenue, et un fonctionnement normal est obtenu par la régulation d'affaiblisseurs de voies de transmission et de réception. Le procédé consiste à déterminer tout d'abord si des niveaux de signaux à réaction vers l'avant ou de retour devraient être utilisés (76). En fonction de cette détermination, les paramètres de signaux et de bruits au niveau de différents points situés le long des voies de transmission et de réception sont évalués (78, 80). En fonction de ces valeurs paramétriques (82), les affaiblisseurs de transmission et de réception sont ajustés afin de maintenir la stabilité de boucle et de faire fonctionner le poste

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15/5,K/1 (Item 1 from file: 348)
DIALOG(R) File 348:EUROPEAN PATENTS
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00498434

HEARING AID.

HORGERAT.

PROTHESE AUDITIVE.

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PATENT (CC, No, Kind, Date): EP 502073 A1 920909 (Basic)
EP 502073 B1 940914
WO 9108654 910613

APPLICATION (CC, No, Date): EP 91900061 901129; WO 90NO178 901129

PRIORITY (CC, No, Date): NO 894806 891130

DESIGNATED STATES: AT; BE; CH; DE; DK; ES; FR; GB; GR; IT; LI; LU; NL; SE

INTERNATIONAL PATENT CLASS: H04R-025/00

CITED PATENTS (WO A): EP 326905 A; EP 335542 A; US 4187413 A; EP 40259 A;
EP 364037 A

NOTE:

No A-document published by EPO

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Lapse: 20000202 B1 Date of lapse of European Patent in a
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19940914, IT 19940914, SE 19941214,

Application: 920909 A1 Published application (A1with Search Report
;A2without Search Report)

Examination: 920909 A1 Date of filing of request for examination:
920620

Examination: 930317 A1 Date of despatch of first examination report:
930102

Grant: 940914 B1 Granted patent

Lapse: 950719 B1 Date of lapse of the European patent in a
Contracting State: SE 941214

Oppn None: 950906 B1 No opposition filed

Lapse: 991020 B1 Date of lapse of European Patent in a
contracting state (Country, date): IT
19940914, SE 19941214,

LANGUAGE (Publication, Procedural, Application): English; English; Norwegian

FULLTEXT AVAILABILITY:

Available Text	Language	Update	Word Count
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CLAIMS B	(English)	EPBBF1	3939
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CLAIMS B	(German)	EPBBF1	3628
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CLAIMS B	(French)	EPBBF1	4316
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SPEC B	(English)	EPBBF1	7734
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Total word count - document A		0
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Total word count - document B		19617
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Total word count - documents A + B		19617
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INTERNATIONAL PATENT CLASS: H04R-025/00

...SPECIFICATION the optimum adaptation of the auditory signal to any hearing residue and simultaneously optimize the hearing aid's response function, hearing aids have been developed wherein the signal processing is performed digitally. The response function is adapted through filtering of the digital signal by means of appropriate filter coefficients, thus permitting the frequency response to some

extent to simulate the response function of a person with normal hearing. If the aids of the digital type are designed as so-called all-in-the-ear aids, the...

...it will be an advantage to have several response curves, in order to adapt the hearing aid's amplification as a function of the frequency to a variety of acoustic environments. It is obvious...

...the range from approximately 1 up to approximately 4 kHz.

Another well-known problem with hearing aids, whether they are digital or analog, is acoustic feedback between sound generator and microphone. Even though the hearing aid is positioned so that it closes the meatus and thus also prevents utilization of any hearing residue, this does not prevent feedback at high amplification, since the sound from the sound generator can be conducted back to the microphone either via the material of the hearing aid or via tissue and bone matter in the vicinity of the meatus. It will therefore be desirable to cancel such a feedback signal, e.g. in connection with the digital signal processing in the hearing aid. As has already been mentioned it is also desirable to utilize any hearing residue at... .

...at least partially open, preferably so that it creates an acoustic transmission channel with a low - pass characteristic between the ear opening and the tympanum. If a channel of this kind is to be used with a hearing aid of the all-in-the-ear type, this makes great demands on the miniaturization of the hearing aid. Moreover, the problem of acoustic feedback will be further accentuated and will need to be...the ear opening to the inner meatus.

This acoustic transmission channel ATC functions as a low - pass filter whose characteristics in reality depend on the volume of the channel and the volume of...

...converted to an analog output signal $s(\text{sub}(r))$ which is smoothed in the reconstruction filter 14. The output signal from the reconstruction filter 14 is conveyed to the input terminals of the sound generator SG whose acoustic output...

...case of high amplifications, e.g. over 55 dB, it will therefore be necessary to cancel this feedback signal, which is done preferably by means of a cancellation filter 35 in the digital signal processor DSP. The cancellation is performed in a purely digital manner in the cancellation filter 35 which can be provided in various ways in the digital signal processor, e.g. in a feedback loop between the output from the equalizer 34 and the input of the compressor 33 as...

...for detection and signal processing in accordance with the invention involving the use of a hearing aid with two microphones will now be described in more detail with reference to figs. 5a...

...dB, in case the amplified microphone signal has a higher level than this. The deconvolution filter 13a gives the signal $s(\text{sub } 1)$ an upper critical frequency of 8 kHz, thereby acting as a band stop, after which the signal $s(\text{sub } 1)$ is transmitted to a first input of the...

...viz. the impedance converter 10b, the microphone amplifier 11b, the compressor 12b and the deconvolution filter 13b to a second input of the sample-and-hold circuit SH with equal band limitation.

By means of a not shown monostable multivibrator MVM the signal $s(\text{sub } 2)$ is...

L Number	Hits	Search Text	DB	Time stamp
1	62	381/71.14.ccls.	USPAT	2003/06/27 13:47
2	26	381/71.14.ccls. and (bpf or bandpass or band adj3 pass)	USPAT	2003/06/27 14:03
3	51	381/94.1-94.9.ccls. and adaptive\$2 and (bpf or bandpass or band adj3 pass)	USPAT	2003/06/27 14:04



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No.	Doccode	Number of pages
1	CTNF	11
2	1449	4
3	892	1

Total number of pages: 16

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